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AUDITORY DETECTION AND SOUND LOCALIZATION FOR COMPUTER-GENERATED INDIVIDUAL COMBATANTS

by

John C. Michaud

June 2005

Thesis Advisor: Jeff Crowson
Second Reader: Donovan Phillips

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AUDITORY DETECTION AND SOUND LOCALIZATION FOR COMPUTER-GENERATED INDIVIDUAL COMBATANTS

John C. Michaud
Captain, United States Army
B.S., United States Military Academy, 1995

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June 2005

Author: John C. Michaud

Approved by: Jeff Crowson
Thesis Advisor

Donovan Phillips
Second Reader

James N. Eagle
Chairman, Department of Operations Research

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ABSTRACT

Soldiers rely predominantly on vision to detect targets, yet other senses may cue their sense of sight. Contrarily, most army combat simulations employ only visual cues. The focus of this thesis is to enhance combat simulations by providing a method by which computer-generated entities can detect and locate objects via a phenomenon known as "sound localization." The Auditory Detection Program is used to represent a human's hearing, and data from a sound localization experiment are analyzed to determine how to best represent the event in which an individual hears a sound and then estimates the location of the sound's source. The resulting algorithms are coded into the Army's combat simulation, $COMBAT^{XXI}$, and the "face-validation" method is used to determine if the algorithms enhance the realism of the simulation. The data analysis consists of Shapiro-Wilks Tests for Normaility, Friedman's Tests for Randomized Block Experiment, and Wilcoxon Rank-Sum Tests using the Bonferroni Correction. Implementing this model in $COMBAT^{XXI}$ improves the simulation by making it more realistic.

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TABLE OF CONTENTS

LIST OF FIGURES

LIST OF TABLES

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LIST OF ACRONYMS AND ABBREVIATIONS

ADM	Auditory Detection Program
ADRPM	Acoustic Detection Range Prediction Model
AFRL/HEC	Air Force Research Lab/Human Effectiveness Crew
AHAAH	Auditory Hazard Assessment Algorithm
ALF	Auditory Localization Facility
ANOVA	Analysis of Variance
ARL	Army Research Lab
CASTFOREM	Combined Arms and Task Force Evaluation Model
CCH	Computer Controlled Hostilities
CGF	Combat Generated Forces
cgs Rayl	A unit of sound impedance that uses the centimeter, gram, and second as its basic units.
CSA	Chief of Staff of the Army
COMBAT^{XXI}	Combined Arms Analysis Tool for the 21st Century
dB	Decibel
DMSO	Defense Modeling and Simulation Office
DoD	Department of Defense
ES2	"Every Soldier is a Sensor"
HTI	Head Tracking Instrument
Hz	Hertz
IAD	Inter-aural Amplitude Difference
IFF	Identification, Friend or Foe
ITD	Inter-aural Time Delay
IWARS	Infantry Warrior Simulation
JND	Just Noticeable Difference
LED	Light Emitting Diode
Leq	Equivalent Sound Level
MAWG	Modeling and Analysis Working Group

MCCDC	Marine Corps Combat Development Command
MOE	Measures of Effectiveness
M&S	Modeling and Simulation
NPS	Naval Postgraduate School
NS	No Sound
OneSAF	The program office that was responsible for creating one Semi-Automated Forces (SAF) Computer Generated Forces
OTB	OneSAF Testbed Baseline
SA	Situational Awareness
SAF	Semi-Automated Forces
SEES	Security Exercise Evaluation System
SEM	Sound with Errors While Moving
SEP	Sound with Errors While Pausing
SME	Subject Matter Expert
SNM	Sound with No Errors While Moving
SNP	Sound with No Errors While Pausing
S-PLUS	S-PLUS Statistical Software Version 6.2
TA	Target Acquisition
TACOM	Tank and Automotive Command
TRAC	TRADOC Analysis Center
TRAC-Monterey	TRADOC Analysis Center at Monterey, California
TRAC-WSMR	TRADOC Analysis Center at White Sands Missile Range
TRADOC	Training and Doctrine Command
TTS	Temporary Threshold Shift
UCCATS	Urban Combat Computer Assisted Training System
V&V	Verification and Validation
VV&A	Verification, Validation, and Accreditation
WAV file	A file format for storing digital audio (waveform) data. Sometimes called WAVE or a Wave file.

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EXECUTIVE SUMMARY

To represent the individual Soldier more accurately in computer simulations, the U.S. Army is improving current computer combat models and developing new models. This modeling and simulation (M&S) initiative is designed to facilitate the Army's transition to the Future Force by allowing leaders to evaluate different force organizations and equipment schemes. To assist this effort, the Training and Doctrine Command Analysis Center (TRAC) – Monterey is currently implementing the Soldier Representation in M&S Master Plan, which emphasizes modeling "the actions, behaviors, and information available to the Soldier." [Ref 21] This thesis directly supports the work done by TRAC – Monterey.

All current Army combat models use vision as a Soldier's primary means of detection. In fact, the detection event causes the computer-generated entities to react to other entities or objects it encounters. Few, if any, of the Army's combat models use any of the other four human senses—hearing, touch, smell, and taste, for detection and reaction. In reality, vision is the dominant sense a Soldier employs to detect objects; however, a Soldier often uses other senses as a cue to react. The focus of this thesis is to enhance combat simulations by providing a method by which computer-generated entities can use their sense of hearing to detect and to locate objects through a phenomenon known as "sound localization." More specifically, this thesis provides combat modelers with a program that robustly models sound propagation and auditory detection, a program that extracts a sound signature for any sound recorded in a WAV file format, and algorithms that replicate a human's sound-localization abilities.

The Auditory Detection Program was selected to model sound propagation and to represent a human's auditory detection capability. The Auditory Hazard Assessment Algorithm was chosen to extract signatures from sound recordings for use in combat simulations. Laboratory data from a sound-localization experiment were analyzed to determine how to represent the phenomenon in which an individual hears a sound and then estimates the location of the sound's source. Data from the Auditory Detection Program, Auditory Hazard Assessment Algorithm, and sound-localization algorithms were then combined and coded into the Army's combat simulation, COMBAT^{XXI}. As a result, COMBAT^{XXI} now provides users with the option to model selected sounds and to create rules that describe how entities react to those sounds. This enhancement has a potential impact on a computer-generated combatant's ability to acquire targets, survive attacks, gather intelligence, and communicate with other entities. The added capability may also allow analysts to examine the use of acoustic sensors and non-lethal acoustic weapons on the battlefield.

One hundred trials were run in COMBAT^{XXI} with the new algorithms and scenarios specifically developed to test the new model. The S-PLUS version 6.2 was used to analyze the data. [Ref 11] The data analysis consisted of Shapiro-Wilks Tests for Normaility, Friedman’s Tests for Randomized Block Experiment, and Wilcoxon Rank-Sum Tests using the Bonferroni Correction. The analysis demonstrated that COMBAT^{XXI} executed with the addition of auditory-detection capabilities and sound-localization algorithms result in statistically different outcomes than COMBAT^{XXI} without the additions.

The “face validation” method was selected to determine whether this model met the requirement for validity. Implementing this model in COMBAT^{XXI} made the computer-generated entities in this simulation act in a more realistic manner.

I. INTRODUCTION

In accordance with guidance from the Chief of Staff of the Army (CSA), Army agencies are "implementing the concept of 'every Soldier is a sensor' (ES2)." [Ref 12] To help analyze this concept, the modeling and simulation community must represent a human's sensory capability in combat simulations more realistically. Currently, most combat simulations use only one of the senses, namely "sight," when modeling humans.

This thesis helps the U.S. Army modeling and simulation community represent the individual Soldier more accurately in combat simulations by providing computer-generated entities with the ability to locate objects via sound. This ability is known as "sound localization." Specifically, this thesis enhances combat simulations with three products: a robust sound-propagation and auditory-detection program, a program that extracts the signature of any sound recorded in a WAV file format, and sound-localization algorithms based upon experiments with human subjects. This more accurate modeling of the individual Soldier enables better analysis of force structures, Soldier systems, tactics, and other Soldier issues.

Consider how important the senses are to a Soldier's life. A loud noise can cause an immediate reaction, prompting one to assume a protective posture, while turning to locate the source of the sound. An individual may feel the vibration of an approaching tank or thud of an artillery round seconds before actually hearing or seeing the object. This provides valuable time for the Soldier to take cover. A Soldier on patrol can smell the smoke of a cigarette, which alerts him to someone's presence, or can smell the stench of chemical weapons, which provokes him to don a protective mask. The added awareness that all the human senses provide is very important in close-combat situations, such as in an urban environment.

Even though algorithms, which can represent human hearing, have existed since the early 1990's, the limitations of computers have discouraged modelers from routinely using algorithms in their simulations. [D. A. Reece, personal communication, December 1, 2004 and Ref 17] Moreover, the line-of-sight calculation for a computer-generated combatant's vision has historically been the most computationally expensive algorithm in computer models. Adding just one of the other senses places another costly burden on the simulation, which can be more expensive than the line-of-sight calculation. [Ref 18] Yet with recent advancements in computer technology, it may be advantageous to include one or more additional human senses in limited situations, such as in small-scale engagements or in situations in which computer generated entities detect only a small number of events via a particular sense.

A. BACKGROUND

The Training and Doctrine Command Analysis Center – White Sands Missile Range (TRAC-WSMR) convened the Soldier Modeling and Analysis Working Group (MAWG) from September 2003 to March 2004 in order to "develop a plan of action to guide future development and use of modeling and simulation (M&S) to support Soldier and small unit decision issues." [Ref 23] The result of this meeting was the "Soldier Modeling and Analysis Working Group Evaluation Report," which identified 56 capability gaps in current combat models.

Many of the capability gaps listed in the MAWG report may be addressed by more accurately modeling the sensor capabilities (senses) of computer-generated entities. Some of the gaps that could be improved include situational awareness (SA), target acquisition (TA), communication, the identification of friend or foe (IFF), and Human Factors. More robust representation of the human senses in computer models may also allow the M&S community to use combat simulations to evaluate the CSA's concept that "every Soldier is a sensor."

B. SCOPE

The purpose for this thesis is to enhance combat simulations by providing a method by which computer-generated entities can detect and locate objects via a phenomenon known as "sound localization." This purpose is achieved by providing three products: a robust sound propagation and auditory detection program, a program that extracts the signature of any sound recorded in a WAV file format, and sound-localization algorithms based upon experiments with human subjects. This thesis focuses on six areas: a study of previous work done with auditory cues in combat simulations and sound-localization experiments, examination of the auditory-detection and sound-localization phenomena, description and analysis of sound-localization data conducted with human subjects, description of the models used to add the auditory-detection capability, and implementation of the models into the Army's combat simulation, COMBAT^{XXI}.

1. Previous Work

a. Auditory Cues in Combat Simulations

Combat simulations have used sound cuing since the early 1990s. Specific examples are the OneSAF and UCCATS simulations. These simulations employed very simple sound-propagation and auditory-detection models. This thesis provides a more robust sound-propagation and auditory-detection program that was developed by auditory experts at the Army Research Laboratory.

b. Sound Localization

Human sound-localization experiments have typically focused on replicating auditory cues with headphones to motivate a human to a desired response. Conversely, this thesis uses data from these experiments to replicate the human's response to an auditory stimulus.

2. Auditory Detection and Sound Localization

a. Auditory Detection

Auditory detection is a function of the physical characteristics of a sound and the individual characteristics of a human. Sound characteristics include frequency, amplitude, temporal characteristics, and location. Human characteristics consist primarily of an individual's hearing threshold. This thesis provides a sound propagation and auditory detection model that accounts for most of these properties.

b. Sound Localization

Sound localization studies show that a subject's directional localization error depends on the location of the sound source in relation to the listener's head. Sounds originating from behind and above the subject yielded errors with greater standard deviations than sounds originating in front of the subject near the horizon. This thesis provides algorithms that produce sound-localization errors consistent with these studies.

3. Localization Data

Raw data from a sound localization experiment, conducted by the U.S. Air Force Research Laboratory, were analyzed to determine how to best replicate a human's ability to use the sense of hearing to ascertain the correct direction of a sound source. A regression tree was used to classify localization errors possessing similar standard deviations with respect to the azimuth and elevation of the sound source that produced the error. Various normal distributions were then selected to represent each group of localization errors in algorithms for use in the in the simulation.

4. The Models

The Auditory Detection Program was chosen for this work because it models both sound propagation and human hearing. The Auditory Hazard Assessment Algorithm augments the Auditory Detection Program by providing sound signatures for any sound recorded in a WAV file format. When calculating sound attenuation, the Auditory Detection Program accounts for the effects of distance between the sound source and the listener, as well as temperature, humidity, "type of ground surface" (grass, sand, snow, etc.), foliage, barriers, winds, and other atmospheric conditions. The auditory-detection calculations consider a listener's hearing threshold and efficiency and the background noise surrounding the listener. The combined detailed calculations of sound propagation and auditory detection make this a highly robust model, which can replicate many diverse situations.

5. Implementation

The sound-localization algorithms, Auditory Detection Program, and Auditory Hazard Assessment Algorithm were implemented in the Army's combat simulation, $COMBAT^{XXI}$, to determine if their combination better models the individual Soldier when compared against $COMBAT^{XXI}$ alone. Analysis of data produced by the combined models showed that they significantly enhanced the realism of the combat simulation.

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II. PREVIOUS WORK

A. AUDITORY CUES IN COMBAT SIMULATIONS

Using sound to cue entities in combat simulations is not new. In 1992, Neta and Mansager investigated the utility of developing sound cuing for the Army's Janus combat simulation. In 1995, Reece developed a sound-cuing method for a Combat Generated Forces (CGF) system called "Computer Controlled Hostiles" (CCH) for the Team Target Engagement Simulator. This sound-cuing method was later implemented in OneSAF Testbed Baseline (OTB) in 2001. In 2003, Balogh created a simple sound-cuing system for use in COMBAT^{XXI}. This system provided rudimentary distance estimation.

Neta and Mansager based their work on the sound-cuing algorithm used in the Urban Combat Computer Assisted Training System (UCCATS). However, this algorithm only used the distance between the listener and the sound source to determine sound attenuation and did not account for atmospheric conditions, terrain, foliage, etc. The authors acknowledged the deterioration of the response time of Janus by implementing a sound-cuing model and recommended applying a simple version of the algorithm on serial computers. If more detailed algorithms were used, the authors recommended using parallel computers and the INTEL iPSC/2 hypercube, available from the Naval Postgraduate School (NPS). [Ref 17] Their report did not detail how sound localization was implemented in this model.

Reece developed a sound-cuing system for use in CCH to counter the unrealistic event in which computer-generated combatants were killed by enemy personnel who walked up behind them. This algorithm allowed entities to hear the footsteps of an enemy approaching them from the rear so they could turn and defend themselves. [D.A Reece, personal correspondence, December 1, 2004] Reece later expanded the sound cuing to include vehicle noises and weapon fire. [Ref 18] Reece's algorithm used the inverse square law to calculate sound propagation and did not account for atmospheric absorption, barriers, foliage, etc. The algorithm provided perfect localization information and fused visual and audio cues to allow an entity to form a situational awareness of threats. [D.A Reece, personal correspondence, December 1, 2004, Ref 18]

Balogh created a simple sound-detection model in COMBAT^{XXI} to support thesis work on modeling how individual entities react to indirect fire. The sound-detection algorithm provided an entity with general distance, but not direction. Given the event that an artillery round exploded on the battlefield, an entity could determine whether the sound was very close, close, near, nearby, or far from its location. [Ref 19] This algorithm did not model sound propagation and did not provide any directional localization information.

Research done thus far with auditory cues in combat simulations has primarily used a simple inverse square law to model sound propagation and has provided the computer-generated entities with perfect localization information. This thesis provides combat modelers with a robust sound propagation and auditory detection model and a method that represents realistic human sound-localization abilities.

B. SOUND LOCALIZATION

Much of the sound-localization research has been conducted to allow humans to use their natural sound-localization abilities with headphones. Locating sound with headphones may provide combat pilots and air traffic controllers with greater situational awareness. These headphones may also provide more realistic conditions for individuals using various training simulators. The general findings of these studies are found in Chapter III, Section B (page 12).

Sound-localization researchers have typically focused their work on understanding how humans localize sounds so the researchers could replicate auditory cues to prompt humans to a desired response. This thesis examines how to model the human reaction to sound cues rather than to replicate the sound cues.

III. AUDITORY DETECTION AND SOUND LOCALIZATION

A. AUDITORY DETECTION

1. Factors

The stimulus for the human auditory experience, hearing, is sound. Sound is "a vibration (actually compression and refraction) of the air molecules," which is often represented mathematically as a sine wave. Hearing is dependent upon four factors of sound: amplitude, frequency, temporal characteristics, and location. [Ref 25]

a. Amplitude

The amplitude of the sine wave is associated with intensity, or loudness and is measured in decibels (dB). The point at which a sound becomes just audible for a person with good hearing is typically zero dB, but this depends upon the frequency of the sound. [Ref 20] Figure 1 combines familiar sounds with the decibel scale for reference.

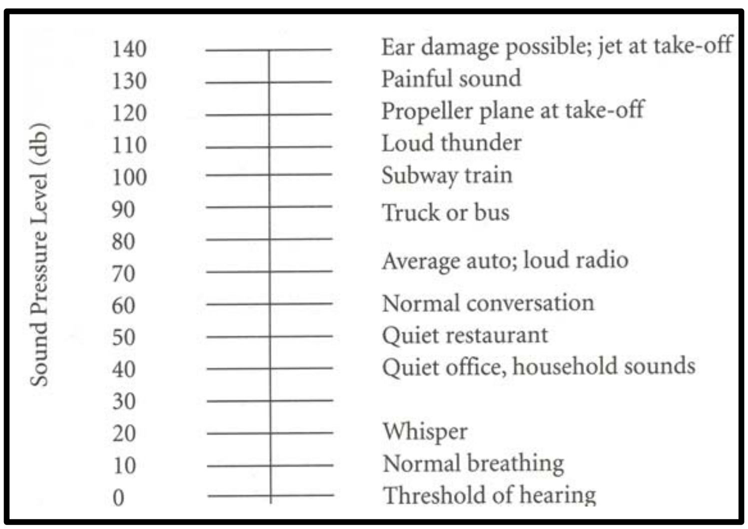

Figure 1. The Decibel Scale [Ref 25]

b. Frequency

The frequency of the sine wave is interpreted as pitch and is measured in cycles per second, or Hertz (Hz). Humans hear in the range from 20 to 20,000 Hz, but are most sensitive to sounds in the 4,000 Hz range. [Ref 25]

Sounds are often described by their energy distribution across the frequency spectrum using octave bands as a reference. Figure 2 provides the standard octave bands used by most scales. A one-third octave band divides the frequency spectrum into three bands per octave. A one-tenth octave band divides the frequency spectrum into ten bands per octave. [Ref 20]

Center Frequency (Hz)	Effective Band (Hz)		
31.5	22.1	to	44.2
63.0	44.2	to	88.4
125.0	88.4	to	177.0
250.0	177.0	to	354.0
500.0	354.0	to	707.0
1,000.0	707.0	to	1,414.0
2,000.0	1,414.0	to	2,828.0
4,000.0	2,828.0	to	5,657.0
8,000.0	5,657.0	to	11,314.0

Figure 2. Standard Octave Bands [Ref 20]

c. Temporal Characteristics

Temporal characteristics include the envelope and rhythm of a sound. The envelope is the set of frequencies of a sound, and the rhythm is the pattern or flow of a sound through time. Temporal characteristics determine the quality of a sound and are important for classification. [Ref 25]

d. Location

Location is the direction and distance of the sound source from the listener. Location will be discussed in detail in Section B of this chapter (page 12).

2. Hearing Threshold

The hearing threshold is the minimum intensity at which a sound can be detected for a specified frequency. [Ref 25] The lowest curve on the graph in Figure 3 shows the hearing threshold in decibels across the frequency spectrum for a person with normal hearing. There are personal thresholds that define the auditory detection ability for people with different levels of hearing loss. "In clinical tests on human subjects, signal intensities slightly above a personal threshold level are always heard, while signal intensities slightly below the threshold are never heard, thus sound detection with changes in intensity . . . approximates an all or none phenomenon." [Ref 5]

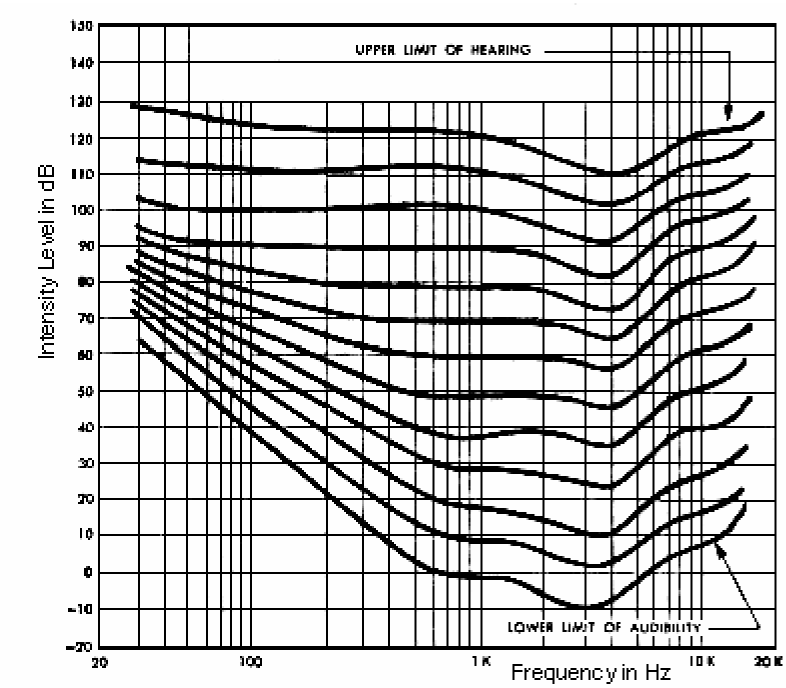

Figure 3. Equal Loudness Contours [Ref 20]

3. Masking

Sounds can be masked by other sounds. Masking is a very complex issue but can be simplified with a few assumptions. Typically, a masking sound must be 6 dB louder than another sound to drown it out totally. This rough estimate is based upon the assumption that the just noticeable difference (JND) in sound intensity for a human is approximately 1 dB and is further based on the fact that two sounds with similar frequencies produce a combined intensity slightly higher than their individual intensities. In order to raise the combined intensity of the two sounds above the JND of 1 dB, and therefore to be detectable, a sound must be within 6 dB of the masking sound. [Ref 16]

B. SOUND LOCALIZATION

Auditory localization is the "recognition of a sounds location." [Ref 3] The auditory system is not as precise as the visual system in determining the location of an object: however, the auditory system has 360 degrees of range while vision has only about 130 degrees of range. [Refs 14,15] People often use their sense of hearing to give them a general idea of a sound's location so they can then use their vision to obtain a more exact position. Localization consists of three components: horizontal, vertical, and distance.

1. Horizontal Localization

Horizontal localization is primarily based upon the Interaural Time Delay (ITD) and Interaural Amplitude Difference (IAD). ITD "is the elapsed time from the incidence of a wave front at the entrance of one ear canal until the same wave front reaches the other ear canal." IAD is the difference between pressure intensity that a stimulus created in one ear from the pressure intensity by the same stimulus created in the other ear. [Ref 3]

Most studies show that a person's best horizontal localization occurs when sounds originate from the front of the head near the horizon. Performance tends to decrease as the sound moves from the front toward the back of the head, with the worst performance consistently to the rear at high or low elevations. [Refs 2, 14] The deterioration of performance lessens when the sound is directly to a subject's side. [Ref 14] One study provides slightly different results. This study shows that horizontal performance is the best directly to the sides of the head, slightly worse to the front, and worst at the rear. [Ref 26] Although results of studies slightly differ for errors produced at the front or at the side of the head, all studies agree that localization is worst for sounds to the rear of a subject at high or low elevations.

2. Vertical Localization

Vertical localization uses the same cues as horizontal localization but is less accurate and has greater error variability. [Refs 14, 26] Vertical localization performance tends to be best in the front of the head near the horizon and decreases with lower and higher elevations and near the back of the head. [Refs 2, 14, 26]

3. Distance

A person's ability to estimate distance via hearing is much less accurate than the ability to estimate angular direction. People often underestimate the distance to faraway sources and overestimate distances of sources less than one meter. Zahorick suggests that compressive power functions provide good approximations for an individual listener's perceived distance. [Ref 28]

The compressive power function is of the form $r' = kr^a$, where r' is an estimate of perceived distance, r is the physical source distance, and k and a are fit parameters that account for an individual listener's ability and the environmental conditions. The parameter a ranges from 0.15 to 0.70, but is on average 0.40. The fitted constant value, k, has an average value slightly greater than one. [Ref 28]

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IV. LOCALIZATION DATA

A. EXPERIMENT DESCRIPTION

The sound localization data used for this thesis was provided by the U.S. Air Force Research Lab's Human Effectiveness: Crew Systems Interface Division (AFRL/HEC). [Refs 1, 4] The following describes the experiment that produced the data:

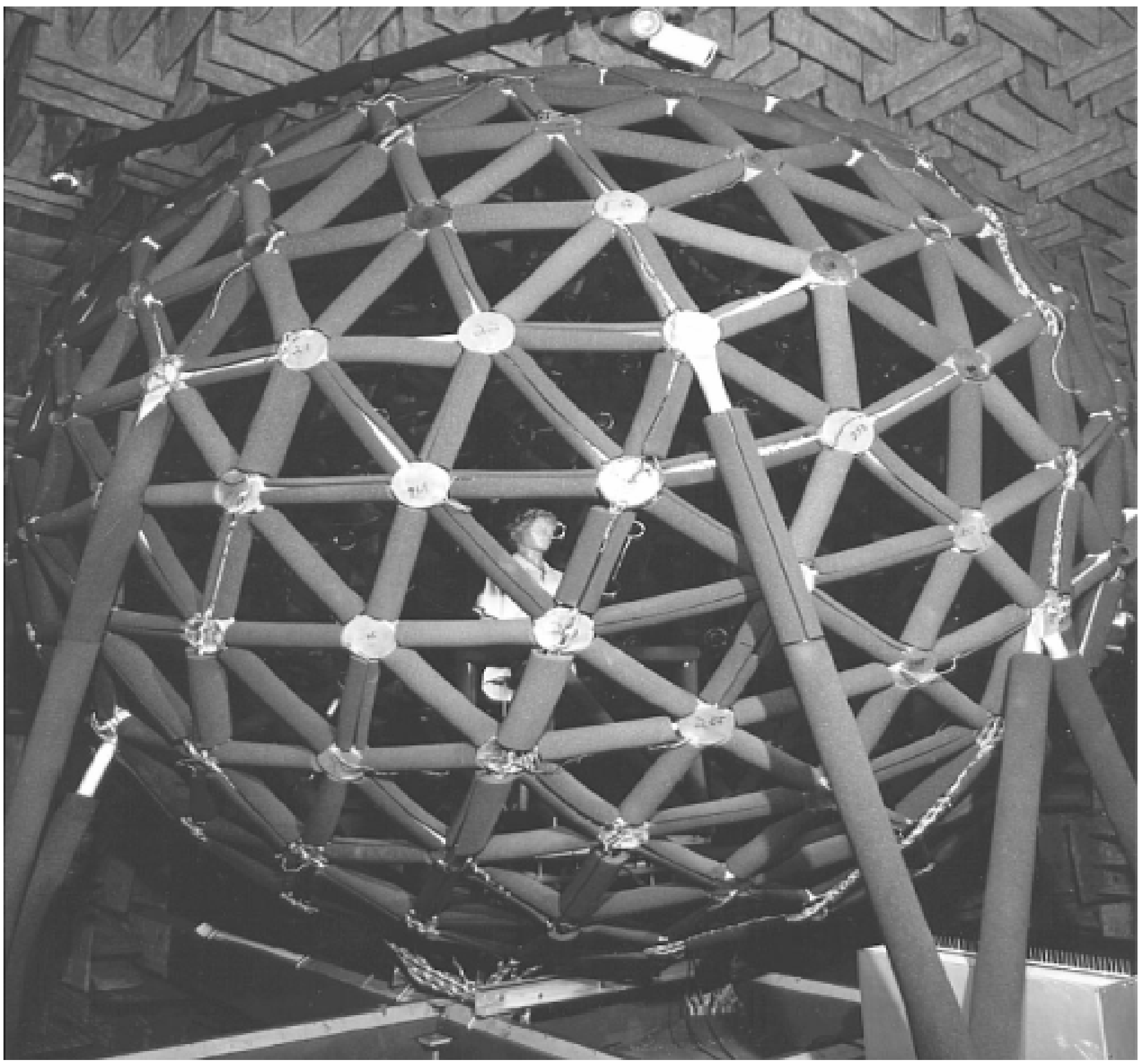

Figure 4. Auditory Localization Facility (ALF) [Ref 4]

The experiment was conducted in the AFRL Auditory Localization Facility (ALF). [See Figure 4] The ALF is a 6.7 m x 6.7 m x 6.7 m anechoic chamber containing an aluminum-frame geodesic sphere, 4.6 m in diameter, with loudspeakers located approximately at each of its 272 vertices. Each of the 272 speaker locations has also equipped with a cluster of four independently-controllable red LEDs. The speakers and LEDs directly faced the listener, who entered the facility through a folding door and stands on a platform that was adjusted in height to make his or her ears level with the sphere's horizontal plane. The subject's head movement was tracked using a Polhemus 3-Space FastTrack. Data from the head-tracker was then fed into the control computer via a Tucker-Davis HTI and transformed into raw azimuth and elevation data. From this point, the computer's azimuth and elevation data determined at which speaker the subject was looking, and the resulting LED cluster on the selected speaker was illuminated providing immediate feedback. The resultant technique allowed the subjects to actively select the speaker from whom they thought the sound originated simply by looking at it. To confirm a response, subjects used a response button, and the LEDs on the selected speaker flickered to provide confirmation. No feedback, however, was provided regarding the correctness of the response. After each response, subjects were required to look at the 0° azimuth 0° elevation speaker and press the response button. This ensured that the subjects were looking at the same point at the beginning of every trial. [A. J. Kordik, personal communication, December 16, 2004]

B. DATA SET DESCRIPTION

1. Information in the Data Set

The data set provided by AFRL/HEC consists of the following information: [A. J. Kordik, personal communication, December 16, 2004 and Ref 1]

- **Subject Number**: Each subject was assigned a number for identification purposes.
- **Trial Number**: The number of the trial for the given subject.
- **Source Speaker Number**: The number assigned to the speaker that produced the audio signal.

- **Source Azimuth**: The azimuth of the speaker that produced the audio signal, measured in degrees from the front of the head. See Figure 5.
- **Source Elevation**: The elevation of the speaker that produced the audio signal, measured in degrees from horizontal. See Figure 6.
- **Level**: The intensity of the sound used in the experiment. The sound level for all trials was 55 dB.
- **Duration Code**: 1 indicates a 250ms signal burst; 2 indicates the stimulus was on continuously until the subject responded.
- **Speaker Code**: 1 indicates that a speaker was chosen randomly from a set of speakers on the horizon; 2 indicates that the speaker was chosen randomly from any elevation above -45°.
- **Open Code**: 1 indicates the subject was not wearing hearing protection, which was true for all trials in this experiment.
- **Timestamp**: The time at which each trial was conducted. Used for record-keeping purposes.
- **Response Speaker Number**: The number assigned to the speaker to which the subject indicated the signal originated.
- **Response Azimuth**: The azimuth measured in degrees from the front of the head where the subjects indicated that they heard the signal noise. Response azimuth uses the same reference system as source azimuth.
- **Response Elevation**: The elevation measured in degrees from the horizontal where the subjects indicated that they heard the signal noise. The response elevation uses the same reference system as the source elevation.

- **Response Time**: The time elapsed, in seconds, from the time the signal noise was played until the subject responded.
- **Hit**: Based upon the subject's response. 1 indicates a perfect response (no error); 0 indicates an imperfect response (with error).
- **Angular Error**: Based upon the subject's response. Combines both the error in azimuth and error in elevation into one value.

2. Reference Systems

a. Horizontal Reference System

The reference system used for azimuth establishes 0° as being directly in front of the subject's head, +90° directly to the right of the head, -90° directly to the left of the head and ±180° directly to the back of the head.

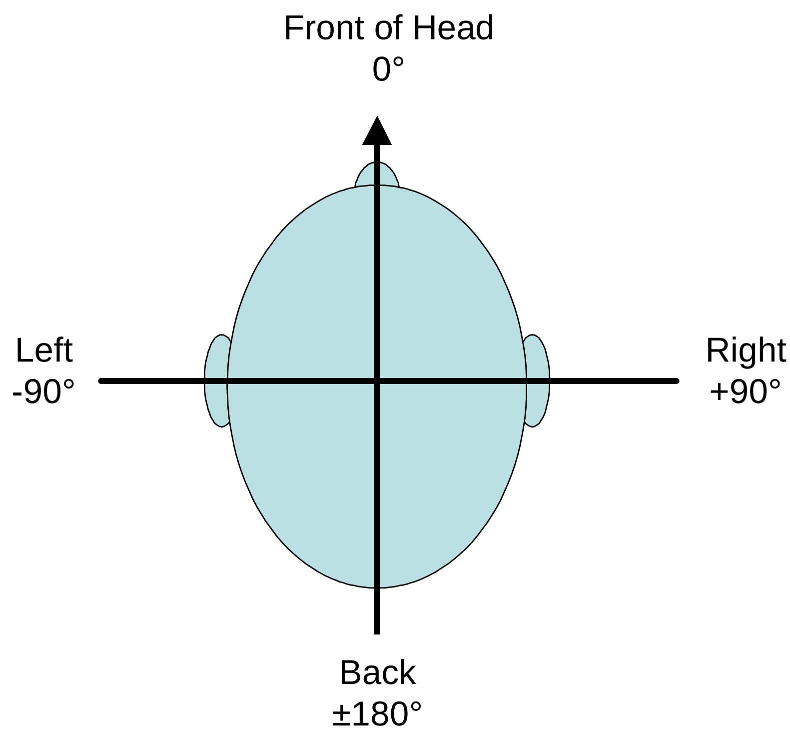

Figure 5. Horizontal Reference System

b. Vertical Reference System

The reference system for elevation establishes the horizontal as 0°. Positive numbers represent elevations above horizontal, and negative numbers represent below the horizon. The values range from -44.7° to 82.4°.

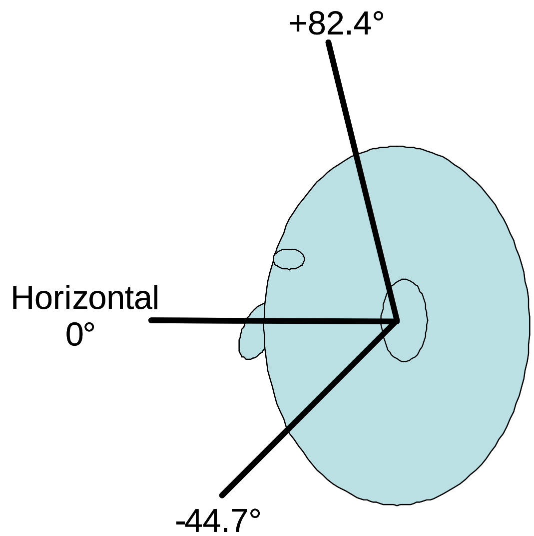

Figure 6. Vertical Reference System (not to scale)

3. Variables

a. Independent Variables

The independent variables for this study are source azimuth and source elevation.

b. Dependent Variables

The dependent variables for this study are response azimuth and response elevation. More specifically, the focus is on azimuth error and elevation error. The azimuth error is calculated by subtracting the source azimuth from the response azimuth. A positive value means the response was to the right of the source azimuth. The elevation error is calculated by subtracting the source elevation from the response elevation. Positive values represent responses higher than the corresponding source elevations.

C. DATA ANALYSIS

The results from this data analysis support the findings listed in Chapter II. Localization is better for sounds in front of the listener near the horizon and becomes worse as the sound approaches the back of the head and/or approaches high or low elevations. Approximately 70% of the responses provided by the subjects were direct hits, which mean they have no error. The remaining 30% have errors, which are represented in the model by various normal distributions.

1. Azimuth Errors

The data was first sorted according to individual speakers. Then data for speakers on the left side of the head were combined with data for speakers at the same location on the right side of the head to obtain a larger sample size for each general speaker location. See Figure 7. In other words, the data for the speaker at azimuth 100° and elevation 0° were combined with the data for the speaker at azimuth -100° and elevation 0°. Each speaker had an individual sample size around 60 observations. The combined sample size approximately doubled the number observations for computational purposes. In order to do this, errors on the right side of a person's head were assumed to be from the same distribution as errors on the left side of the head. The standard deviations for each location were calculated and the errors were found to be heteroschedastic. See Figure 8. Areas to the front of the head near the horizon have smaller standard deviations than areas to the back of the head at high or low elevations.

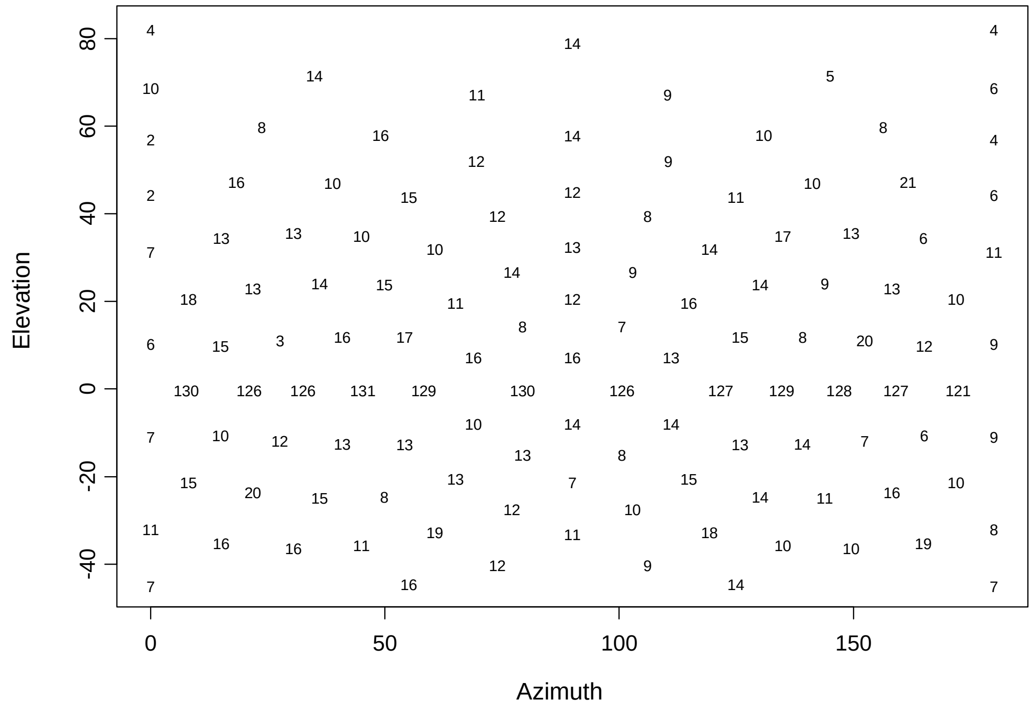

Figure 7. Number of Times Each Location Was Used as a Sound Source

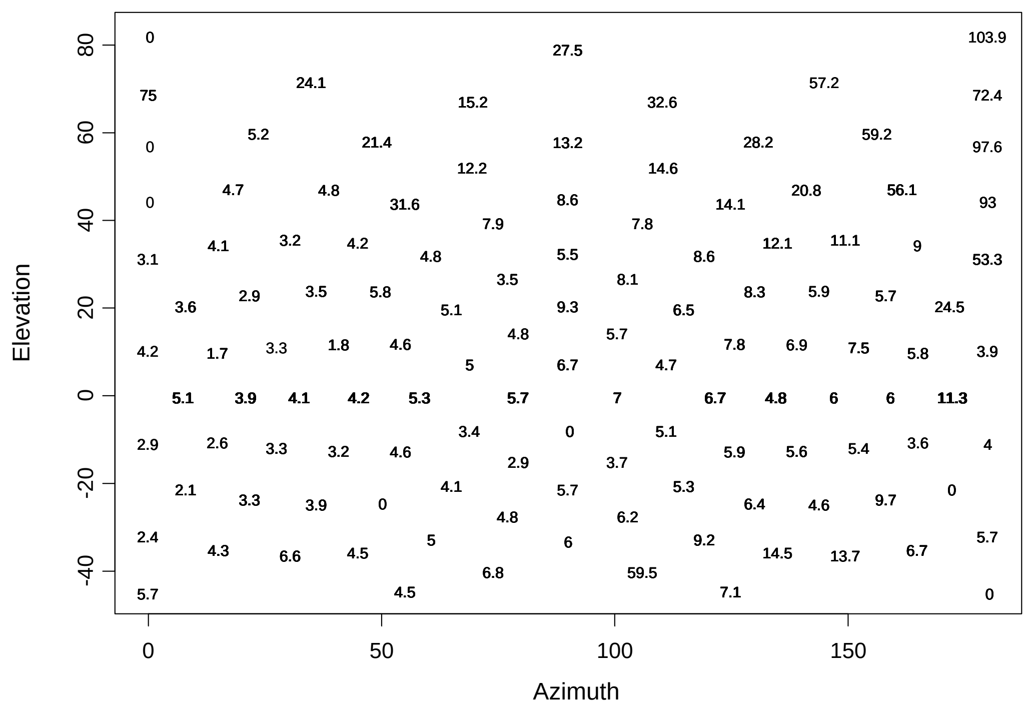

Figure 8. Standard Deviations for Each Speaker Location

Next, the statistical software S-PLUS was used to generate a regression tree to sort the localization errors by standard deviations into similar groups. Figure 9 displays the regression tree, and Figure 10 provides the same information in a different format. Note that src.el stands for “sound source elevation” and src.az stands for “sound source azimuth.”

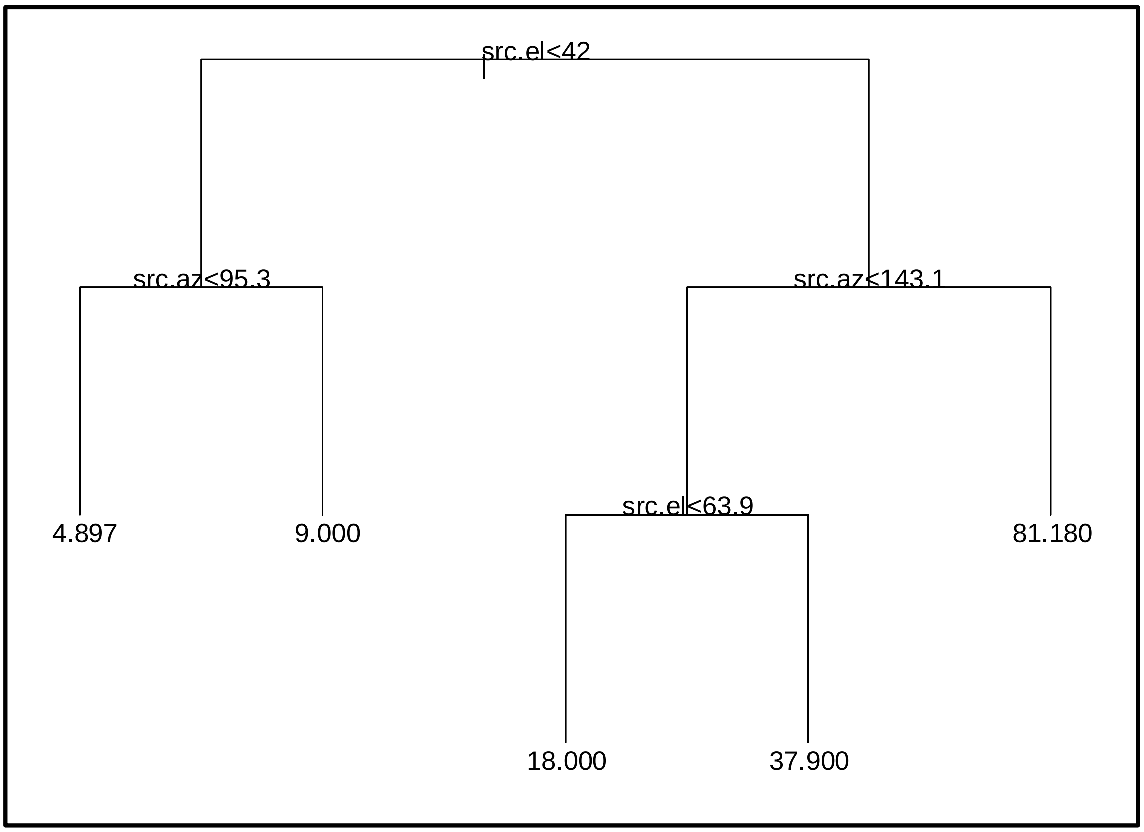

Figure 9. Regression Tree for Azimuth Error Standard Deviations

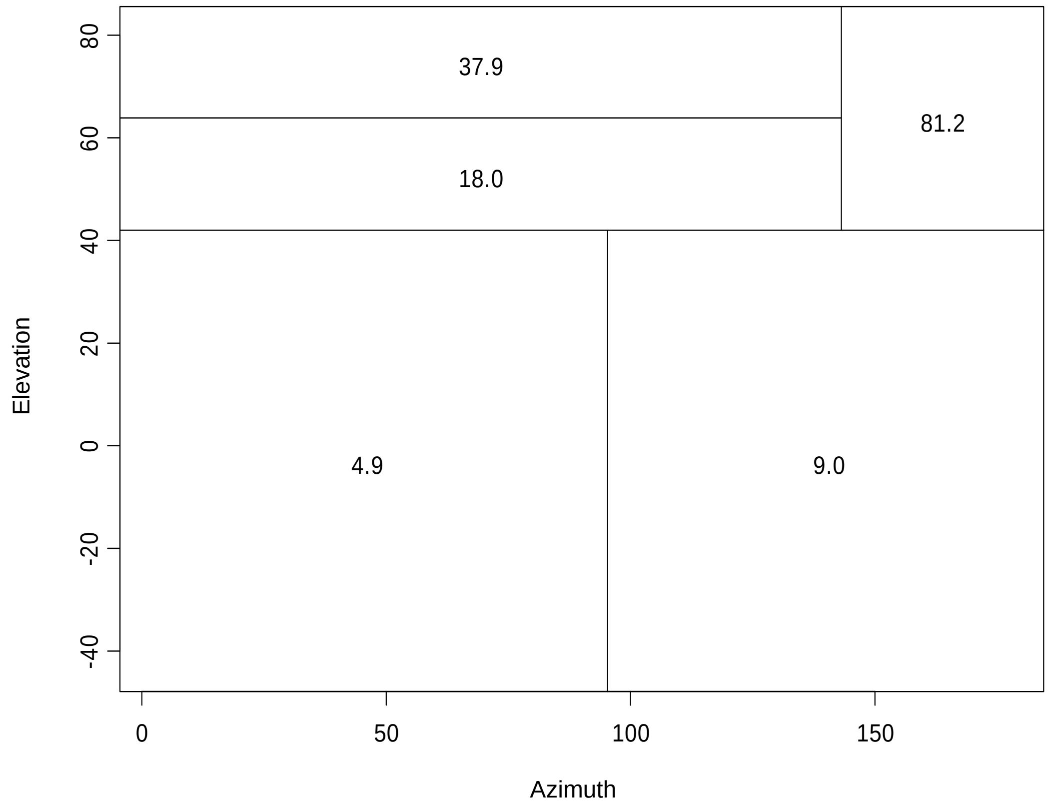

Figure 10. Regression Tree Surface for Standard Deviations

Once the data were divided into regions with the regression tree, the azimuth errors within each region were analyzed and the best method by which to replicate the data for random number generation was determined. First, the Shapiro-Wilks test for normality was used to determine if the complete data sets (sets that include perfect responses) within each region could have come from the normal distribution. This test showed that the complete data sets for all of the regions were significantly different from the normal distribution. The direct hits were then removed from the data and the same test was performed on the remaining data. A few of these data sets were not significantly different from the normal distribution if an alpha of 0.01 was used. For regions with data that remained significantly different from the normal distribution, a combination of various normal distributions was chosen to represent the errors by plotting their histograms and selecting the combinations that produced plots that best matched the histogram for the actual data.

a. *Source Elevation Less Than 42° and Source Azimuth Less Than 95.3°*

There were 1,345 data points, 80% of these were perfect responses. The remaining 20% were represented by a combination of two normal distributions: 10% of the data set with a mean of -11 and a standard deviation of 4, and the final 10% with mean of 11 and standard deviation of 4. Note that the horizontal reference for the histograms below is “degrees of error.”

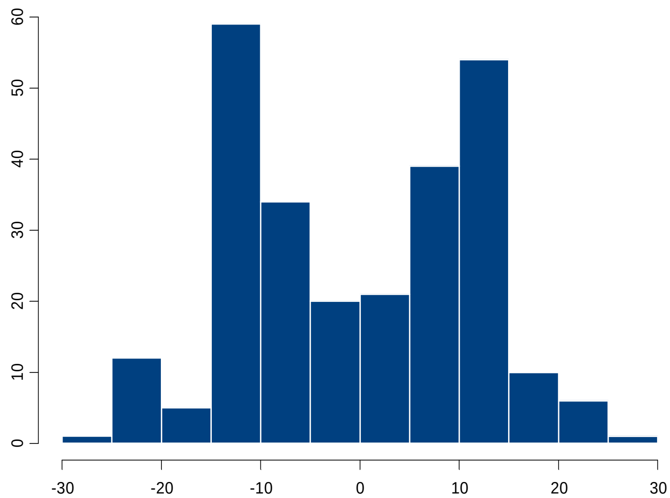

Figure 11. Actual Data: Elevation < 42°, Azimuth < 95.3°

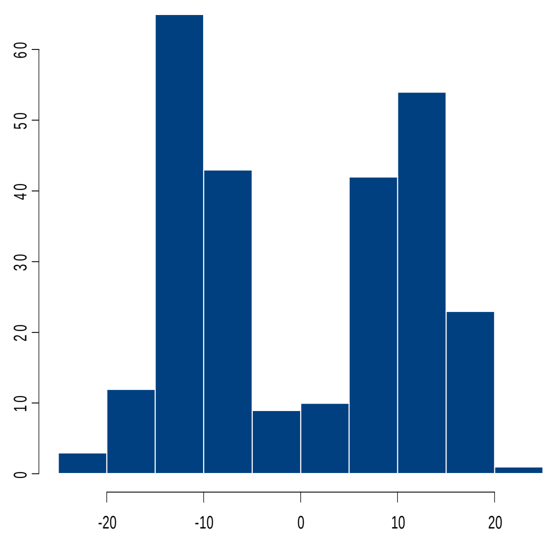

Figure 12. Replicated Data: Elevation < 42°, Azimuth < 95.3°

b. Source Elevation Less Than 42° and Source Azimuth Greater Than 95.3°

There were 1,224 data points, 68% of these were perfect responses. The remaining 32% were represented by a combination of two normal distributions: 31% of the data set with a mean of 42 and a standard deviation of 23, and the final 1% with a mean of 0 and a standard deviation of 100. Note that the horizontal reference for the histograms below is “degrees of error.”

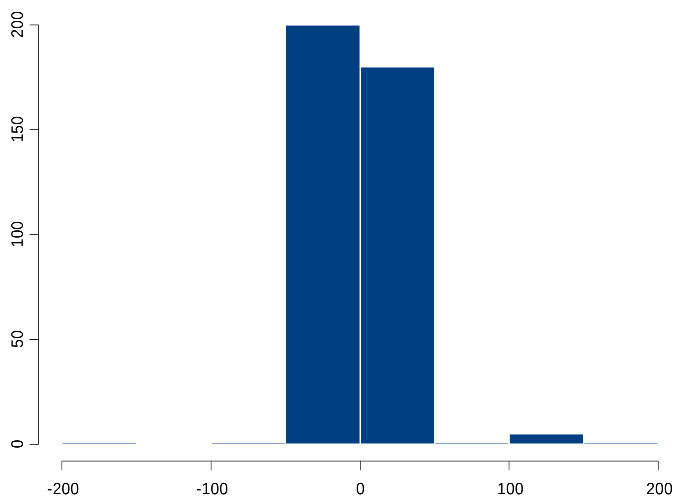

Figure 13. Actual Data: Elevation < 42°, 95.3° ≤ Azimuth

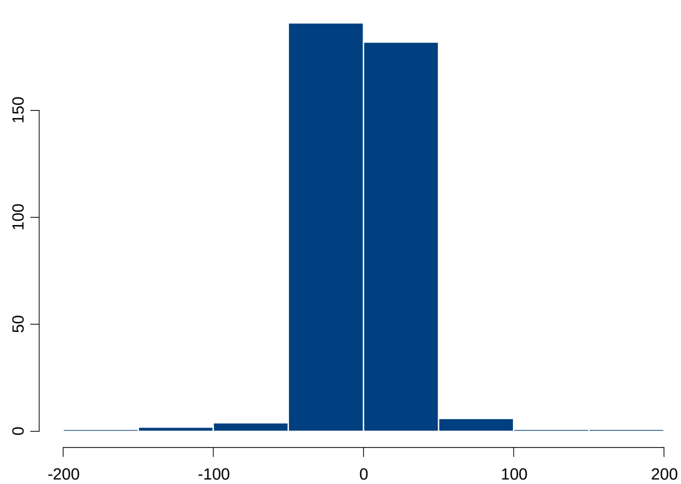

Figure 14. Replicated Data: Elevation < 42°, 95.3° ≤ Azimuth

c. *Source Elevation between 42° and 63.9° and Source Azimuth Less Than 143.1°*

There were 147 data points, 50% of these were perfect responses. A Shapiro-Wilks test performed on the remaining 50% produced a *w*-value of 0.95 and a *p*-value greater than 0.01. Therefore, a normal distribution was used to represent this data with a mean of -4 and a standard deviation of 32.2. The mean and standard deviation were computed from the 50% of the data without perfect responses. Note that the horizontal reference for the histograms below is "degrees of error."

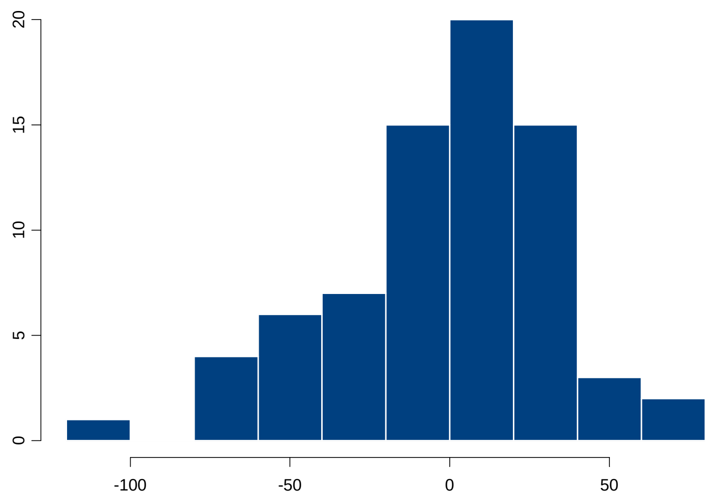

Figure 15. Actual Data: 42° $\leq$ Elevation < 36.9°, Azimuth < 143.1°

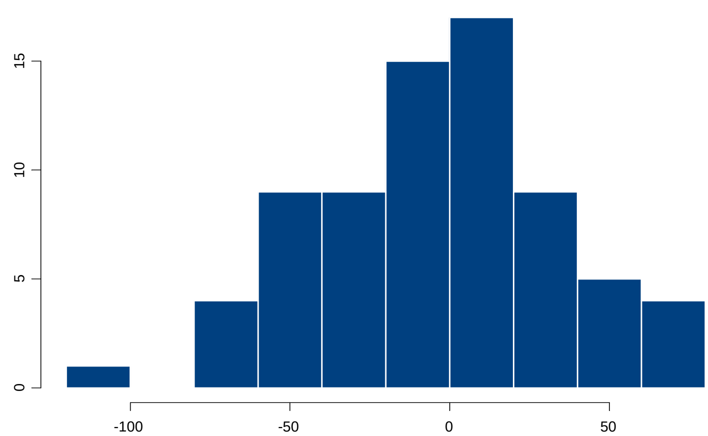

Figure 16. Replicated Data: 42° $\leq$ Elevation < 36.9°, Azimuth < 143.1°

d. Source Elevation Greater Than 63.9° and Source Azimuth Less Than 143.1°

There were 62 data points, 53% of these were perfect responses. A Shapiro-Wilks test performed on the remaining 47% produced a *w*-value of 0.92 and a *p*-value greater than 0.02. Therefore, a normal distribution was used to represent this data with a mean of 26 and a standard deviation of 61. The mean and standard deviation were computed from the 47% of the data without perfect responses. Note that the horizontal reference for the histograms below is “degrees of error.”

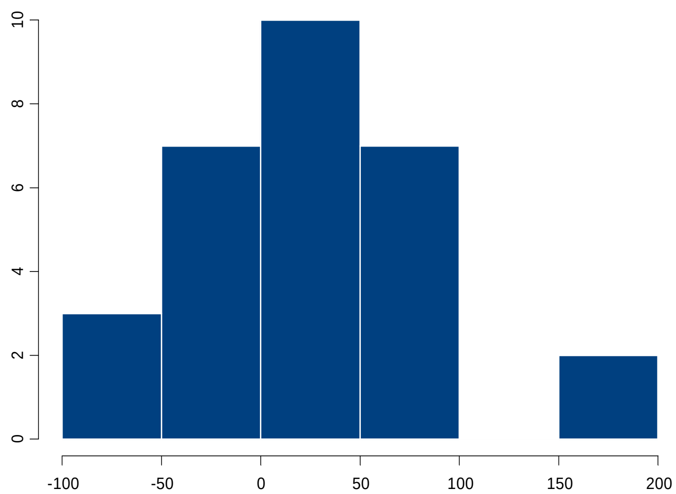

Figure 17. Actual Data: 36.9° ≤ Elevation, Azimuth < 143.1°

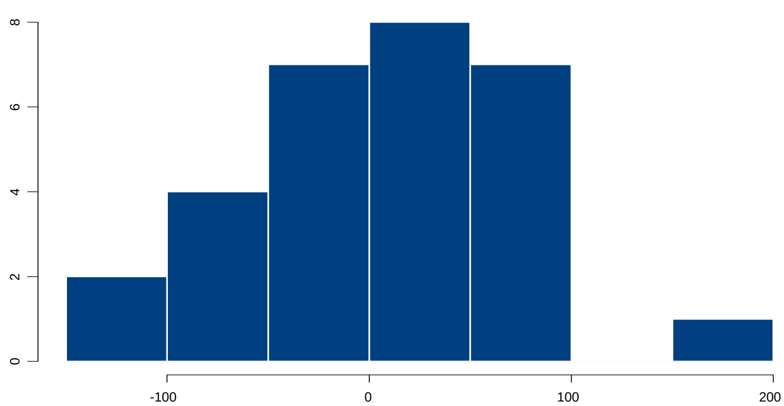

Figure 18. Replicated Data: 36.9° ≤ Elevation, Azimuth < 143.1°

e. Source Elevation Greater Than 63.9° and Source Azimuth Greater Than 143.1°

There were 54 data points, 39% of these were perfect responses. A Shapiro-Wilks test performed on the remaining 61% produced a *w*-value of 0.93 and a *p*-value greater than 0.03. Therefore, a normal distribution was used to represent this data with a mean of -51 and a standard deviation of 103. The mean and standard deviation were computed from the 61% of the data without perfect responses. Note that the horizontal reference for the histograms below is "degrees of error."

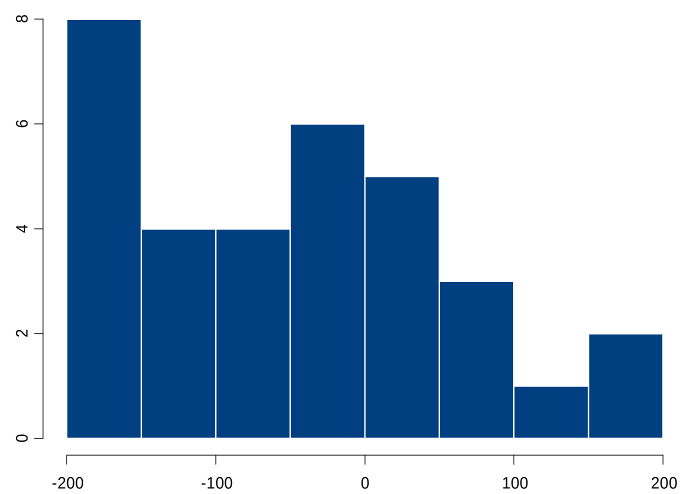

Figure 19. Actual Data: 36.9° ≤ Elevation, 143.1° ≤ Azimuth

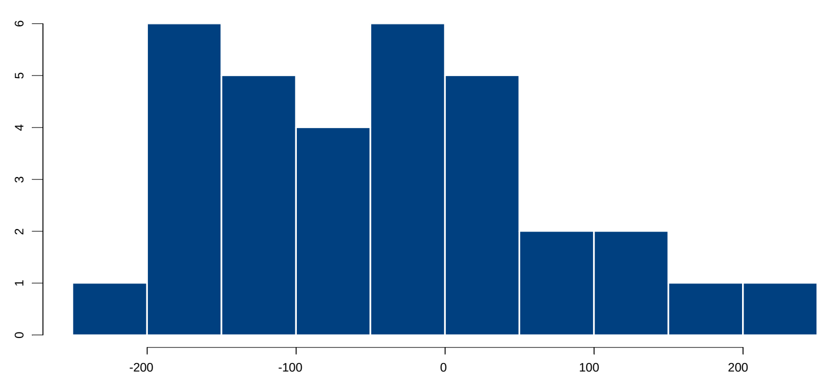

Figure 20. Replicated Data: 36.9° ≤ Elevation, 143.1° ≤ Azimuth

2. Elevation Errors

The correlation coefficient between azimuth errors and elevation errors was calculated to be 0.17. Although this implies a weak positive correlation, independence between the azimuth errors and elevation errors was assumed. Therefore, the same procedure used to determine the best method to replicate azimuth errors was used to determine elevation errors. Note that src.el stands for "sound source elevation" and src.az stands for "sound source azimuth."

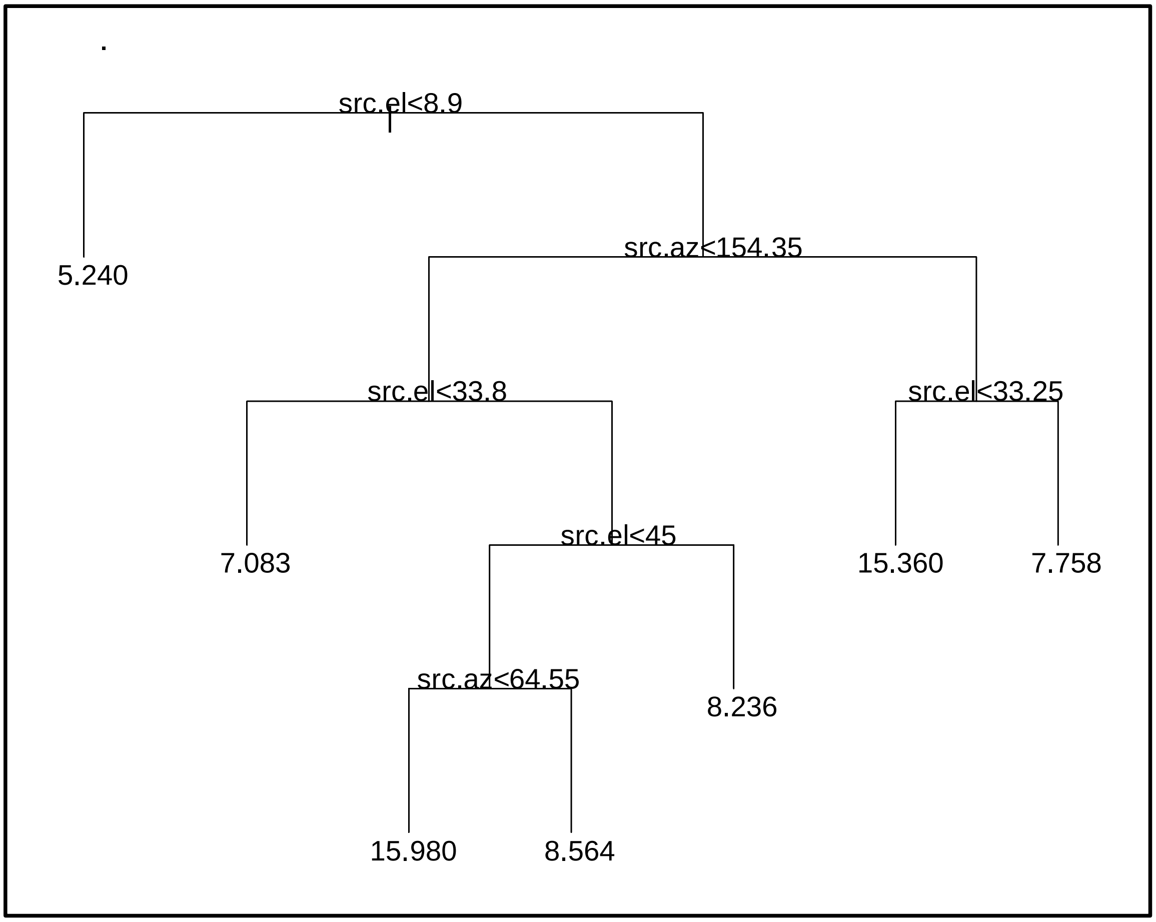

Figure 21. Regression Tree for Elevation Error Standard Deviations

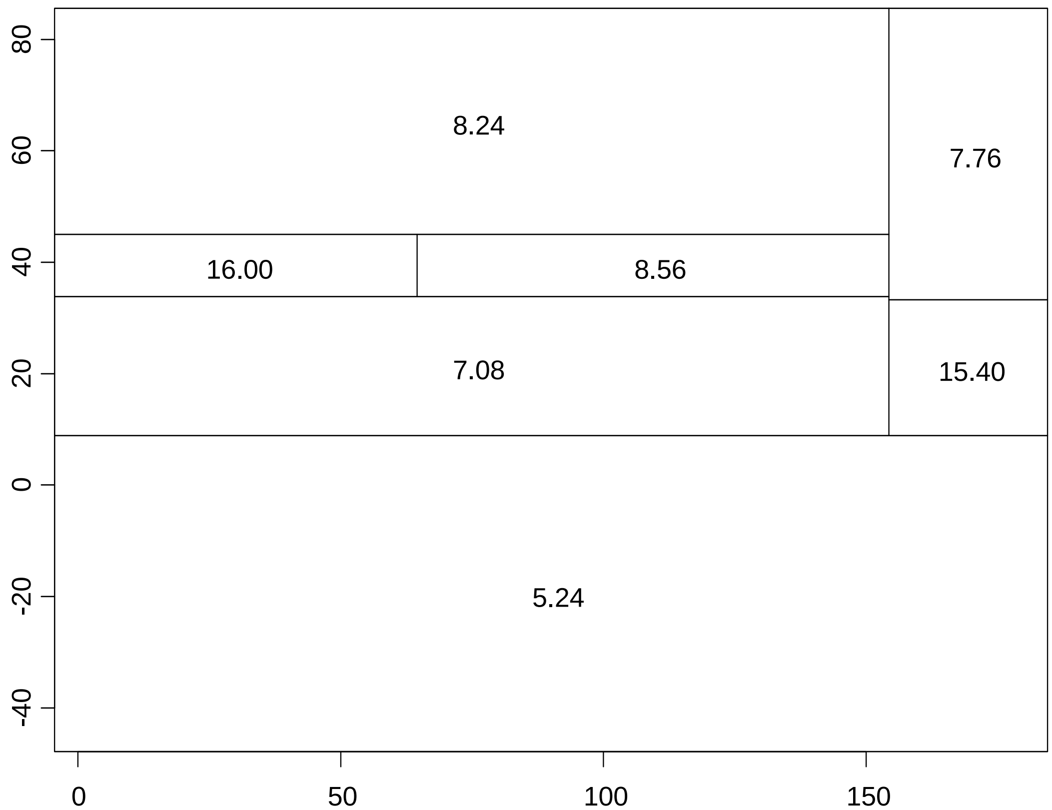

Figure 22. Regression Tree Surface for Elevation Error Standard Deviations

a. Source Elevation Less Than 8.9°

There were 726 data points, 70% of these were perfect responses. The remaining 30% were represented by a combination of three normal distributions: 17% of the data set with a mean of 10 and a standard deviation of 3, 11% with a mean of -10 and a standard deviation of 3, and the final 2% with a mean of 25 and a standard deviation of 15. Note that the horizontal reference for the histograms below is "degrees of error."

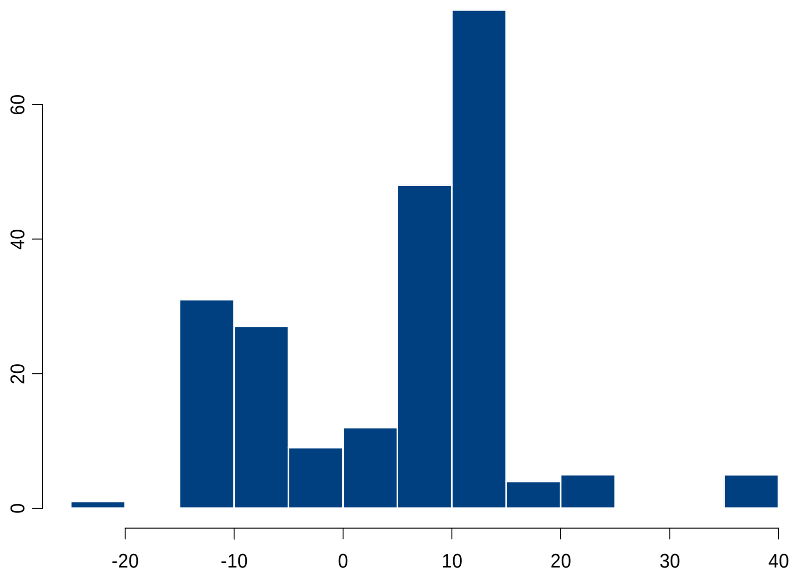

Figure 23. Actual Data: Elevation < 8.9°

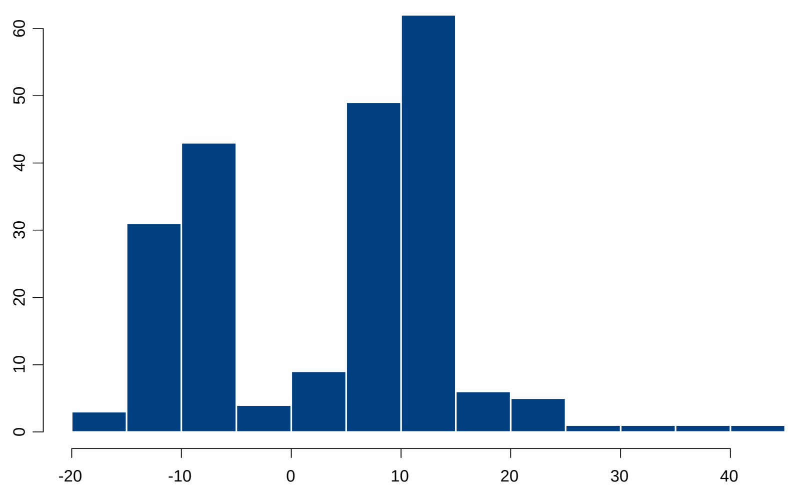

Figure 24. Replicated Data: Elevation < 8.9°

b. Source Elevation between 8.9° and 33.8°, and Azimuth Less Than 154.35°

There were 304 data points, 58% of these were perfect responses. The remaining 42% were represented by a combination of two normal distributions: 25% of the data set with a mean of 10 and a standard deviation of 1, and the final 17% with a mean of 3 and a standard deviation of 10. Note that the horizontal reference for the histograms below is “degrees of error.”

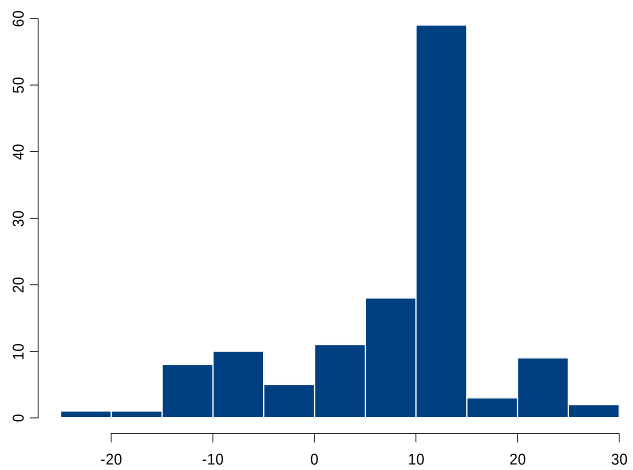

Figure 25. Actual Data: 8.9° $\leq$ Elevation < 33.8°, Azimuth < 154.35°

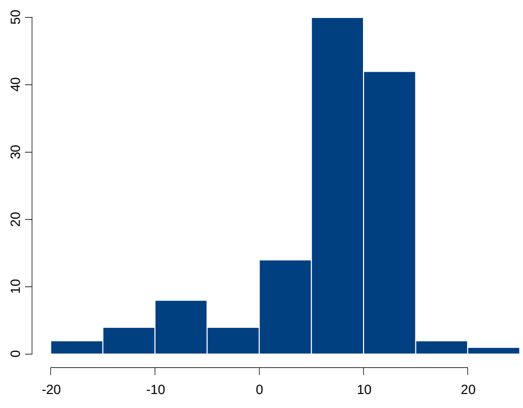

Figure 26. Replicated Data: 8.9° $\leq$ Elevation < 33.8°, Azimuth < 154.35°

c. *Source Elevation between 33.8° and 45°, and Azimuth Less Than 64.55°*

There were 53 data points, 38% of these were perfect responses. The remaining 62% were represented by a combination of two normal distributions: 53% of the data set with a mean of 10 and a standard deviation of 8, and the final 9% with a mean of -60 and a standard deviation of 10. Note that the horizontal reference for the histograms below is “degrees of error.”

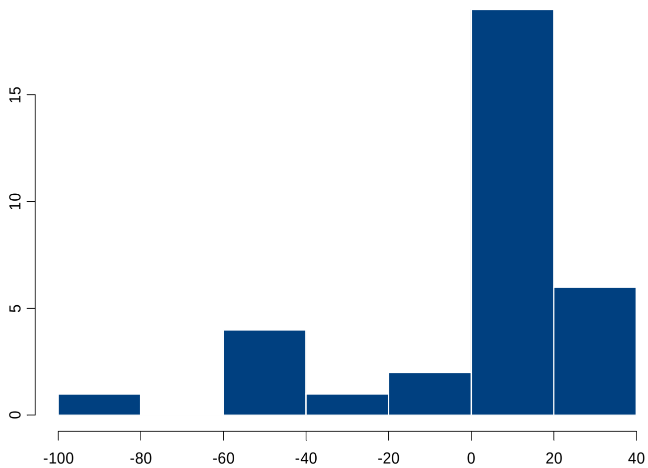

Figure 27. Actual Data: 33.8° ≤ Elevation < 45°, Azimuth < 64.55°

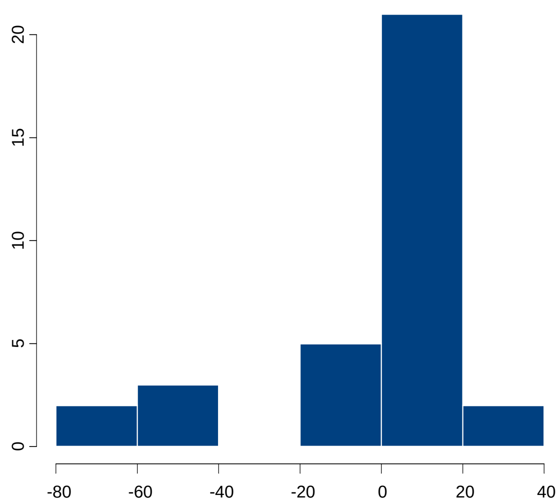

Figure 28. Replicated Data: 33.8° ≤ Elevation < 45°, Azimuth < 64.55°

d. Source Elevation between 33.8° and 45°, and Azimuth between 64.55° and 154.35°

There were 61 data points, 36% of these were perfect responses. A Shapiro-Wilks test performed on the remaining 64% produced a *w*-value of 0.96 and a *p*-value greater than 0.1. Therefore, a normal distribution was used to represent this data with a mean of 7 and a standard deviation of 10. The mean and standard deviation were computed from the 64% of the data without perfect responses. Note that the horizontal reference for the histogram below is “degrees of error.”

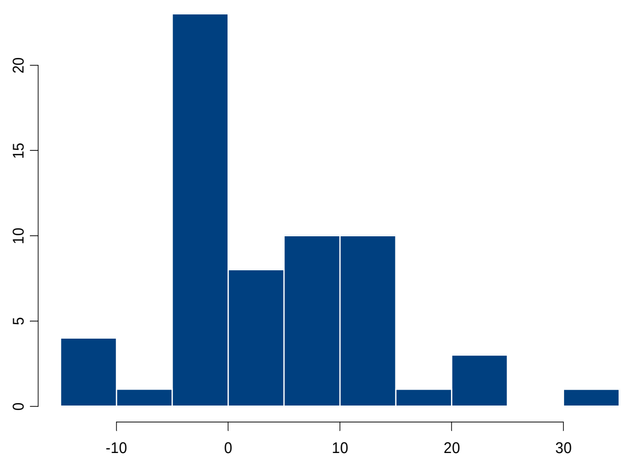

Figure 29. Actual Data: 33.8° ≤ Elevation < 45°, 64.55° ≤ Azimuth < 154.35°

e. ***Source Elevation Greater Than 45°, and Azimuth Less Than 154.35°***

There were 186 data points, 50% of these were perfect responses. The remaining 50% were represented by a combination of two normal distributions: 26% of the data set with a mean of 12 and a standard deviation of 4, and the final 24% with a mean of -10 and a standard deviation of 5. Note that the horizontal reference for the histograms below is “degrees of error.”

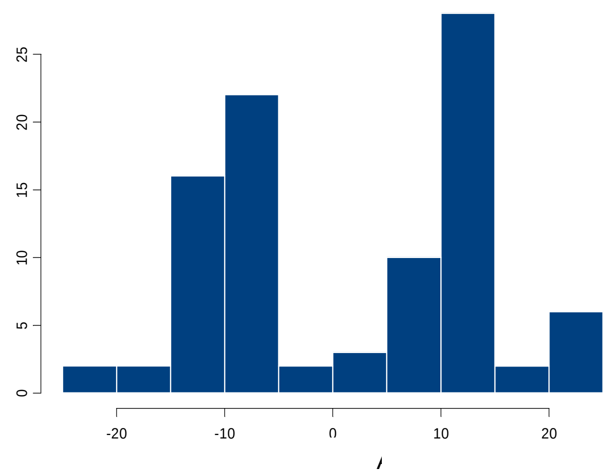

Figure 30. Actual Data: 45° ≤ Elevation, Azimuth < 154.35°

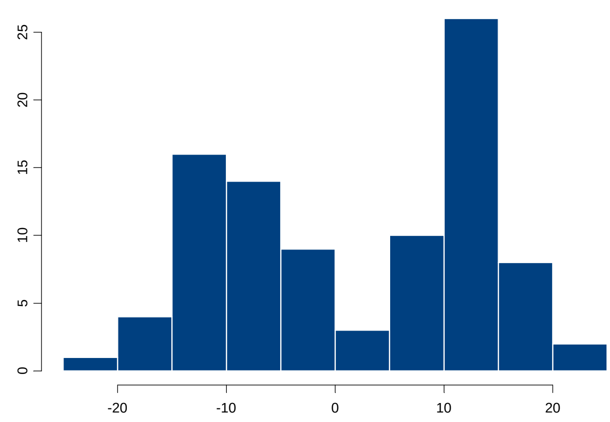

Figure 31. Replicated Data: 45° ≤ Elevation, Azimuth < 154.35°

f. *Source Elevation between 8.9° and 33.25°, and Azimuth Greater Than 154.35°*

There were 55 data points, 36% of these were perfect responses. A Shapiro-Wilks test performed on the remaining 64% produced a *w*-value of 0.92 and a *p*-value greater than 0.01. Therefore, a normal distribution was used to represent this data with a mean of 35 and a standard deviation of 9. The mean and standard deviation were computed from the 64% of the data without perfect responses. Note that the horizontal reference for the histogram below is "degrees of error."

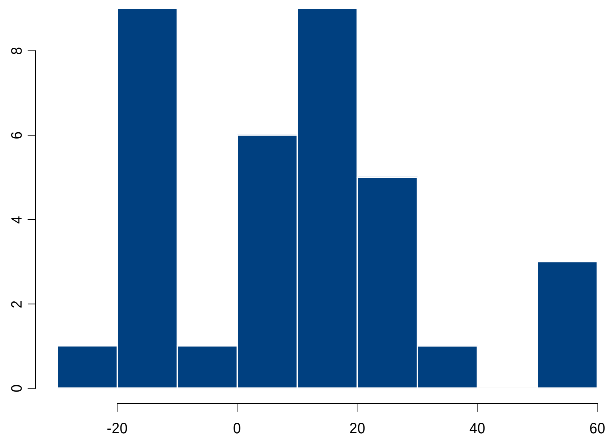

Figure 32. Actual Data: 8.9° ≤ Elevation < 33.25°, 154.35° ≤ Azimuth

g. ***Source Elevation Greater Than 33.25°, and Azimuth Greater Than 154.35°***

There were 55 data points, 40% of these were perfect responses. A Shapiro-Wilks test performed on the remaining 60% produced a *w*-value of 0.94 and a *p*-value greater than 0.05. Therefore, a normal distribution was used to represent this data with a mean of 15 and a standard deviation of 14. The mean and standard deviation were computed from the 60% of the data without perfect responses. Note that the horizontal reference for the histogram below is "degrees of error."

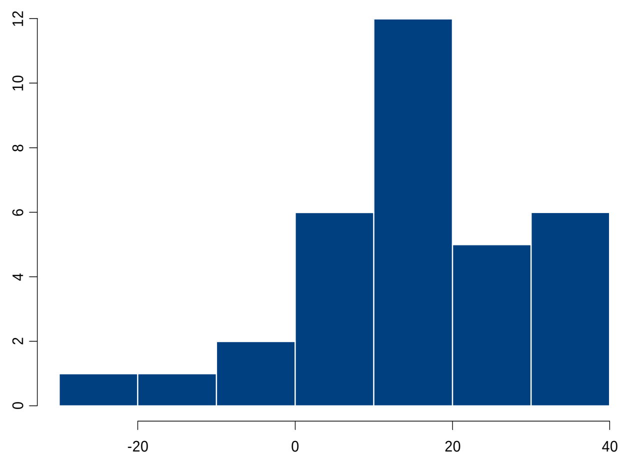

Figure 33. Actual Data: 33.25° $\leq$ Elevation, 154.35° $\leq$ Azimuth

D. DATA USE

The information resulting from the data analysis above is used to replicate human sound-localization errors in a simulation. When a sound is generated, the true direction from the listener to the sound is computed (azimuth and elevation) and compared against the regression-tree surface to see in which region it falls. A random number is then drawn from the distribution that best represents the errors in that region and is applied to the true direction to the sound. This imperfect information is then provided to the entity for use. See Figures 34 and 35.

```
algorithm compute imperfect azimuth;
begin
    calculate the true  azimuth of the sound source from the listener with
        respect to the listener's head
    calculate the true elevation of the sound source from the listener with
        respect to the listener's head
    draw a random number between 0 and 1
    if elevation < 42° and azimuth < 95.3° then
        if rand < 0.8 then
            error = 0.0
        if 0.8 ≤ rand < 0.9 then
            draw error from normal distribution (-11.0, 4.0)
        if 0.9 ≤ rand then
            draw error from normal distribution (11.0, 4.0)
    if elevation < 42° and 95.3° ≤ azimuth then
        if rand < 0.68 then
            error = 0.0
        if 0.68 ≤ rand < 0.99 then
            draw error from normal distribution (42.0, 23.0)
        if 0.99 ≤ rand then
            draw error from normal distribution (0.0, 100.0)
    if 42° ≤ elevation < 63.9° and azimuth < 143.1° then
        if rand < 0.5 then
            error = 0.0
        if 0.5 ≤ rand then
            draw error from normal distribution (-4.0, 32.2)
    If 63.9° ≤ elevation and azimuth < 143.1° then
        if rand < 0.53 then
            error = 0.0
        if 0.53 ≤ rand then
            draw error from normal distribution (26.0, 61.0)
    if 63.9° ≤ elevation and 143.1° ≤ azimuth then
        if rand < 0.39 then
            error = 0.0
        if 0.39 ≤ rand then
            draw error from normal distribution (-51.0, 103.0)
    add the error to the true azimuth
    provide the imperfect azimuth to the entity for use
end;
```

Figure 34. Algorithm to Compute Imperfect Azimuth

```
algorithm compute imperfect elevation;

begin
    calculate the true  azimuth of the sound source from the listener with
        respect to the listener's head
    calculate the true elevation of the sound source from the listener with
        respect to the listener's head
    draw a random number between 0 and 1
    if elevation < 8.9° then
        if rand < 0.7
            error = 0.0
        if 0.7 ≤ rand < 0.87 then
            draw error from normal distribution (10.0, 3.0)
        if 0.87 ≤ rand < 0.98 then
            draw error from normal distribution (-10.0, 3.0)
        if 0.98 ≤ rand then
            draw error from normal distribution (25.0, 15.0)
    if 8.9° ≤ elevation < 33.8° and azimuth < 154.35° then
        if rand < 0.58 then
            error = 0.0
        if 0.58 ≤ rand < 0.83 then
            draw error from normal distribution (10.0, 1.0)
        if 0.83 ≤ rand then
            draw error from normal distribution (3.0, 10.0)
    if 33.8° ≤ elevation < 45.0° and azimuth < 64.55° then
        if rand < 0.38 then
            error = 0.0
        if 0.38 ≤ rand < 0.91 then
            draw error from normal distribution (10.0, 8.0)
        if 0.91 ≤ rand then
            draw error from normal distribution (-60.0, 10.0)
    If 33.8° ≤ elevation < 45.0° and 64.55° ≤ azimuth < 154.35° then
        if rand < 0.36 then
            error = 0.0
        if 0.36 ≤ rand then
            draw error from normal distribution (7.0, 10.0)
    if 45.0° ≤ elevation and azimuth < 154.35° then
        if rand < 0.5 then
            error = 0.0
        if 0.5 ≤ rand < 0.76 then
            draw error from normal distribution (12.0, 4.0)
        if 0.76 ≤ rand then
            draw error from normal distribution (-10.0, 5.0)
    if 8.9° ≤ elevation < 33.25° and 154.35° ≤ azimuth then
        if rand < 0.36 then
            error = 0.0
        if 0.36 ≤ rand then
            draw error from normal distribution (35.0, 9.0)
    if 33.25° ≤ elevation and 154.35° ≤ azimuth then
        if rand < 0.4 then
            error = 0.0
        if 0.4 ≤ rand then
            draw error from normal distribution (15.0, 14.0)
    add the error to the true elevation
    provide the imperfect elevation to the entity for use
```

Figure 35.   Algorithm to Compute Imperfect Elevation

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V. THE MODELS

The algorithms developed in Chapter IV are combined with the three models described in this chapter in a manner that better represents the individual Soldier during combat simulations. The Auditory Detection Program represents sound propagation and human auditory detection. The Auditory Hazard Assessment Algorithm extracts signatures compatible for use with the Auditory Detection Program from any sound recording in a WAV file format. COMBAT^{XXI} is the combat simulation itself. The combination of these algorithms and programs provides the entities in COMBAT^{XXI} with the ability to detect and to locate objects via the sense of hearing.

A. AUDITORY DETECTION PROGRAM

To remain consistent with the documentation for this program, the Auditory Detection Program will be referred to as the ADM. [Refs 8 and 9] Dr. Joel T. Kalb from the Human Research and Engineering Directorate of the U.S. Army Research Laboratory (ARL) provided the ADM for use in this thesis.

1. Model Description

Georges R. Garinther, Joel T. Kalb, David C. Hodge, and G. Richard Price developed the ADM as a tool to revise the aural non-detectability limits for the Department of Defense Criteria Standards for Noise Limits, MIL-STD-1474 (1979). The ADM is based on the Acoustic Detection Range Prediction Model (ADRPM) used by the U.S. Army Tank and Automotive Command (TACOM). The ADRPM "is a software program that models the propagation of acoustic energy through the atmosphere and the detectability of that energy." [Ref 7] The ADRPM was created in the 1970s by the BBN Corporation and has been upgraded though the years. TACOM currently uses it to determine how susceptible mechanized vehicles are to detection by acoustic means. The ADM improves the ADRPM by adding the human detection algorithm and hearing thresholds, as well as including turbulence in the calculations for ground effect. [J. T. Kalb, personal communication, December 12, 2004] Some of the calculations used in the ADM are described in the document, “Proposed aural nondetectability limits for army materiel.” [Ref 8]

> The Auditory Detection Program (ADM) has the capability of computing: The distance at which a target can be detected by unaided human hearing, the one-third octave-band spectrum not to be exceeded for non-detection at any specified distance, and the propagation losses and noise spectrum at any distance from a measured noise source. This model may be broken down into two broad categories: the propagation of sound from the source to the ear of the listener, and the psychoacoustic factors that determine the probability that a listener will hear the sound once it reaches the ear. Both of these categories are controlled by a number of phenomena, the first producing different rates of attenuation at different frequencies, the second influencing the listener's performance and hearing sensitivity. The propagation factors included in the model are geometric spreading, atmospheric absorption, ground effect, refraction due to wind and temperature gradients (this portion of the model is presently being upgraded), barriers, and foliage. The psychoacoustic factors are hearing sensitivity, background noise, and listener performance (This includes the critical bandwidth of human hearing, hit probability, false alarm rate, and listener efficiency). Computing detection distance is accomplished by comparing the sound level of the target calculated for the listener's location when detection occurs to both the listener's threshold of hearing and the background noise level. All measured noise data are in one-third octave-bands. Detection is assumed when the target sound is above both the threshold of hearing and background noise in at least one-third octave-band. [Ref 9]

2. Model Inputs

The model inputs fall into four categories: Sound measurement parameters, detection parameters, listener parameters, and computation parameters.

a. Measurement Parameters

Measurement parameters provide information about the sound signature at the time of recording. These include:

- The distance of the sound source from the microphone in meters. Default is 30 meters.

- The height of the sound source above the ground in meters. Default is 1.2 meters.
- The height of the microphone above the ground in meters. Default is 1.2 meters.
- The ground surface flow resistivity in *cgs* Rayls. The default is 200 *cgs* Rayls. The ranges of possible values for this parameter are found in Table 1.
- The temperature in degrees Celsius. Default is 15° Celsius.
- The relative humidity in percent. The default is 70%.

Ground Surface	**Values for Flow Resistivity (cgs)**
Snow, new fallen	4 to 30
Sand	33 to 40
Forest floor, pine	20 to 80
Grass: rough pasture, airport, institutional, etc.	150 to 300
Roadside dirt, ill-defined, small rocks up to 4 inches.	300 to 800
Sandy silt, hard packed by vehicles	800 to 2500
Old dirt roadway, fine stones (1/4 inch mesh) interstices filled	2,000 to 4,000
Limestone chips, thick layer (1/2 to 1 inch mesh)	1,500 to 4,000
Earth, exposed and rain-packed	4,000 to 8,000
Quarry dust, fine, very hard-packed by vehicles	5,000 to 20,000
Asphalt, sealed by dust and use	> 20,000

Table 1. Range of Values for Ground Surface Flow Resistivity

b. Detection Parameters

Detection parameters provide information about the situation to be modeled in the simulation. These include:

- The height of the sound source in meters. Default is 1.2 meters.
- The height of the listener in meters. Default is 1.2 meters.
- The ground surface flow resistivity in cgs Rayls. Default is 200 cgs Rayls. See Table 1 for a list of possible values.
- The temperature in degrees Celsius. Default is 15° Celsius.
- The relative humidity in percent. Default is 70%.
- The atmospheric condition. This parameter details whether the listener is upwind or downwind of the sound source, the average wind speed in meters per second, the surface temperature in degrees Celsius, the temperature gradient in degrees Celsius per meter, and the gradient layers in meters above the ground (altitude) for each of the following profiles:
    - Neutral Profiles
        - Isothermal
        - Mid-latitude, summer
        - Mid-latitude, winter
        - Tropical, moist
    - Stable Profiles
        - Mid-latitude, summer night
        - Mid-latitude, winter night
        - Desert, summer night
        - Desert, winter night
        - Tropical, moist night
    - Unstable Profiles
        - Mid-latitude, summer day
        - Mid-latitude, winter day
        - Desert, summer day
        - Desert, winter day

- Tropical, moist day

- Information for barriers between the sound source and the listener. Data required are the number of barriers between the sound source and the listener, the distance of the barrier from the sound source in meters, and the height of the barrier in meters.
- Information for foliated areas between the sound source and the listener. Data required are the number of foliated areas between the sound source and the listener, the distance from the sound source to the nearest edge of the foliated area in meters, the depth (extent) of the foliated area in meters, the average leaf width in centimeters, and the average leaf area per unit volume in 1/meters. Typical values are provided in Table 2.

Type of Planting	**Leaf Area per Volume (m^{-1})**	**Average Leaf Width (cm)**
Field of corn	6.3	7.4
Tidal reeds	3.0	3.2
Dense Hardwood Brush	0.5	5.0

Table 2. Typical Values for Foliated Area Parameters

c. Listener Parameters

Listener parameters provide information about the listener to be modeled in the simulation.

- **Listener Efficiency**: The listener efficiency is a value between zero and one, with zero being not efficient and 1 being machine like. Humans are typically 0.4.
- **Hit Probability**: This is the detection probability that ranges between zero and one. A 75% hit probability means that 75% of

listeners will detect the sound event modeled. The default value is 50%.

- **False Alarm Rate:** This value ranges from zero to one. This is the probability that an individual thinks he detects a sound when no sound event occurred. The default value is 0.01%
- **Hearing Threshold:** This is represented as an array of 24 numbers. Each value in the array is the minimum intensity required for an individual with a specified profile to detect a sound. The position of the number in the array corresponds to a frequency in the one-third octave band. See Table 3. The profiles include thresholds for people with the following:
    - Perfect Hearing
    - 1.5 to 2.4 years of military service
    - 7.5 to 12.4 years of military service
    - 17.5 to 22.4 years of military service
    - Poor Hearing
    - Poor Hearing with a Temporary Threshold Shift (TTS).

Temporary Threshold Shift is an event in which an individual's hearing threshold is temporarily degraded due to exposure to a loud noise. An example is when a person experiences ringing in the ears and/or has trouble hearing for a period of time after listening to loud music at a concert.

One-third Octave Band Frequency (Hz)	**Perfect Hearing Threshold (dB)**
50	41.7
63	35.6
80	29.8
100	25.1
125	20.8
160	16.8
200	13.8
250	11.2
315	9.0
400	7.2
500	6.0
630	5.0
800	4.4
1,000	4.2
1,250	3.8
1,600	2.6
2,000	1.0
2,500	-1.2
3,150	-3.5
4,000	-3.9
5,000	-1.1
6,300	6.5
8,000	15.3
10,000	16.4

Table 3. Perfect Hearing Threshold Across the One-third Octave Band

d. Computation Parameters

Computation parameters provide information about the target generating the sound, the background noise, and the precision that the program uses to find the output distance.

- **Target Spectrum:** Like the hearing threshold, this is represented as an array of 24 numbers. Each number represents the intensity of sound produced by the target. The position in the array corresponds to a particular frequency in the one-third octave band. Profiles include an M60 tank idling and an unspecified target created for analysis. Target spectra may be acquired from recordings using the AHAAH. See Chapter V, Section B (page 53) for details.
- **Background Spectrum:** Like the hearing threshold and target spectrum, this is represented as an array of 24 numbers. Each number represents the intensity of sound produced by the background. The position in the array corresponds to a particular frequency in the one-third octave band. Background profiles include, but are not limited to
    - Jungle, day
    - Jungle, night
    - Desert, low wind
    - Desert, moderate wind
    - Urban
    - Rural and Suburban Areas (EPA upper limit)
    - Rural and Suburban Areas (EPA lower limit)
    - North rim of the Grand Canyon (extreme quiet)
    - Noisy jungle.
- **Detection Range Precision:** This is the precision for the line search, conducted by the ADM, to determine the output range. The default value is 0.1%

3. Model Assumptions and Limitations

a. Assumptions

(1) Point Source: The ADM assumes sounds generated are from a point source. A point source is an "idealized sound source, or close actual approximation to it, that radiates sound uniformly in all directions in a free-field situation." [Ref 20] This assumption may cause more detection events to occur for a sound than would happen in real life. This will occur because the sound in the simulation will travel a farther distance in more directions than under actual conditions, and therefore be detected by more people.

(2) Reflective Surfaces: This program does not model sounds bouncing off of reflective surfaces other than the ground. This assumption may cause sounds to propagate farther than they actually would in a real situation because it ignores the dispersion caused by reflected surfaces and the interference of reflected sound waves on the original sound wave (echoes).

(3) Novice Listener: The ADM assumes the listener is not actively listening for sounds. This will result in a smaller number of detections than would probably actually happen. Individuals who are actively listening for a particular sound will probably be able to detect it at a lower intensity.

(4) Detection vs. Classification: The model calculates the point at which a sound can be detected, not necessarily classified. Detection is the event in which a human has heard something, but does not know what was heard. Classification is the event in which a human has heard something and knows what was heard. By treating a detection as a classification, this assumption may cause a larger number of desired reactions to a sound cue than would actually happen. A person who merely detects a sound may attempt to better classify a sound before reacting whereas a person who classifies a sound will react immediately.

(5) Detection Probability: Listeners have a 50% chance of detection if a sound's intensity is equal to their hearing threshold and both the intensity and threshold are greater than any background noise. This assumption should have little effect on the output of the model.

b. Limitations

(1) Sound Signatures: The sound signatures used for this study did not include information describing the conditions during recording. The microphone distance and height during the time of recording for small-arms weapons were assumed to be 0.5 meters and 1.2 meters respectively. The microphone distance was estimated by comparing the maximum intensity level for each rifle signature to the known intensity levels of these weapons, which were measured from the location of the firer's head (about 0.5 meters). [Ref 27] The height is the typical height of a weapon when fired from a standing position. This lack of information should have minimal effect on the output of the model, assuming the sounds were not produced under extreme conditions.

(2) Temperature and Humidity: It was found that the variables that store the temperature and humidity during the time of sound recording were not used anywhere in the code. This should have a minimal effect on the output for the model as long as sounds are not recorded under conditions that vary greatly from the standard temperature and humidity of 15° Celsius and 70% humidity.

4. Model Use

One can see that the ADM accounts for many details in a sound detection event. This robust capability makes the ADM a great tool for use with combat simulations. There are three possible ways to use the ADM: pre-calculate probabilities, employ a cookie-cutter method, and create a new code that calculates the probabilities directly. Due to time constraints and the workload required to implement the other methods, the cookie-cutter method was chosen for immediate use in COMBAT^{XXI}.

a. Pre-calculate Probabilities

In its current form, the ADM calculates the distance at which a percentage of listeners detect a sound. In other words, a user provides a probability, perhaps 75%, and the ADM provides the distance at which 75% of listeners detect the sound. With this method, the simulation uses the ADM to calculate the distances at which 99% and 2% of listeners detect the given sound. These ranges are pre-calculated at the start of the simulation and used throughout the simulation to determine whether or not a listener has detected the sound. For simplification, a linear relationship of the probabilities between the near and far distances is assumed. A detection event is dependent upon the range of the listener from the sound source. Let x be the range of the listener from the sound source, then:

$$P\left\{\text{detection}\right\} = \begin{cases} 1 & \text{if } x \text{ - 99\% range} < 0 \\ \dfrac{0.97(\text{x - 99\% range})}{\text{2\% range} - \text{99\% range}} + 0.99 & \text{if 99\% range} \le x \text{ - 99\% range} \le \text{ 2\% range} \\ 0 & \text{otherwise} \end{cases}.$$

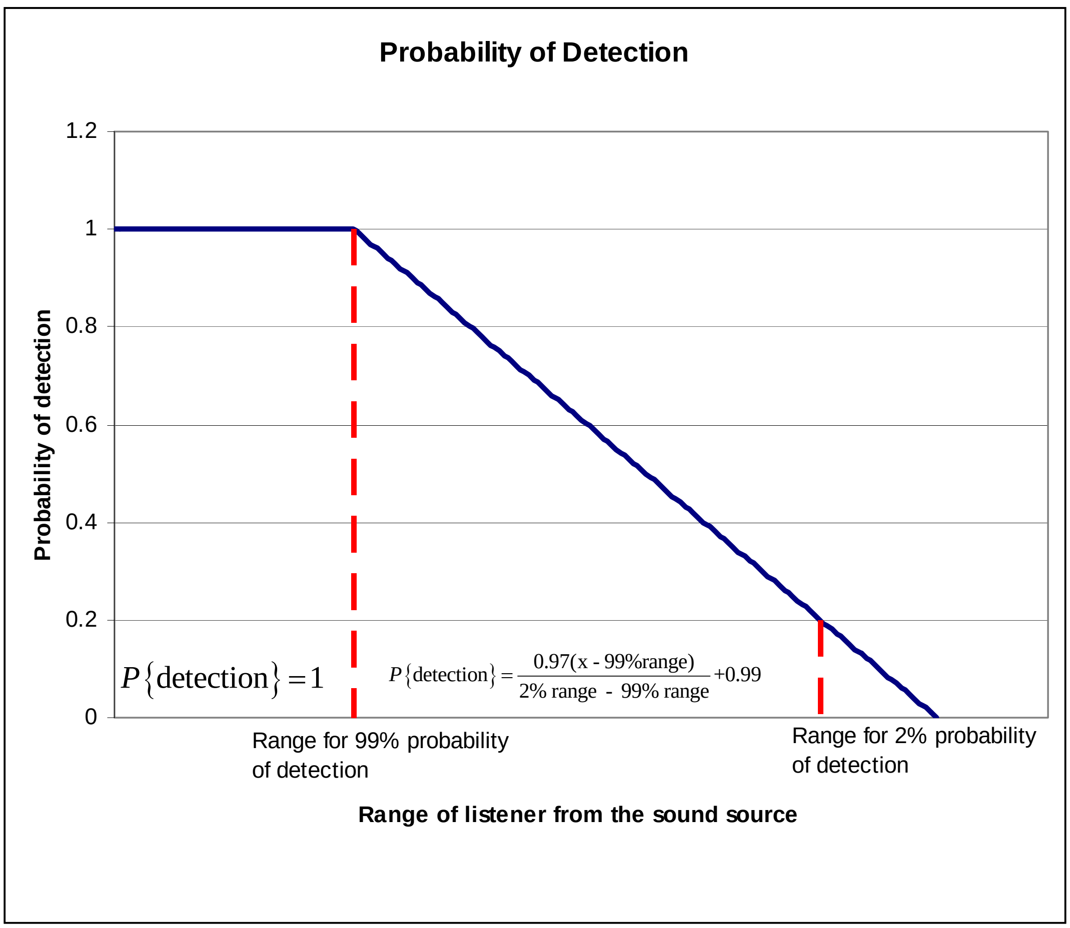

Figure 36. Pre-calculation Method

This method has advantages because it facilitates stochastic modeling by using probabilities, performs the complicated sound propagation calculations at the beginning of the simulation, which saves computational resources, and which has the potential to model a myriad of sounds. The disadvantages are that the method is laborious to code into the combat simulation and does not account for the effects of weather, foliated areas, or barriers that may lie between the listener and the sound source.

b. Cookie Cutter

With this method, the ADM is used separately from the combat simulation to calculate the ranges at which there is a 99% probability of detection for a specified sound. If the listeners are within this range of the sound, they hear it. If they are beyond this range, they do not hear it. This concept is supported by clinical tests on human subjects. These tests show that sound detection with changes in intensity (or distance from the sound source) approximates an all-or-none phenomenon, as just described. [Ref 5]

To use this method, one must assume that all listeners have the same hearing ability, changes in environmental factors have little effect on the propagation of the sound, and a sound occurring beyond an individual's range of influence is inconsequential. In other words, if a Soldier hears rifle fire from three miles away, being outside the effective range of the weapon, the source of the rifle fire is beyond his visual range, and he cannot move to engage the target in a timely manner. Therefore, no evasive action is required and no attempt to acquire the target is warranted. If, however, a Soldier hears a rifle 200 meters away, then he should take action because he is within range of the weapon. Since the distances at which the small arms weapons have a 99% chance of detection are typically large (greater than a mile and beyond the range of an individual's influence), individuals are cued to noises within this range and ignore sounds beyond it.

This method facilitates a simple cookie-cutter detection routine, which is easy to code and which does not require further use of the complicated propagation model. This is the method ultimately implemented in COMBAT[XXI].

c. Create New Code

It may be possible to rewrite the ADM code to calculate a detection probability directly given the range of the listener from the sound source rather than estimating the probability as described in Part "A" above. This method would allow for stochastic modeling and would be the most accurate of the methods listed because it could use the current states of a listener, the sound, and the environment to determine the detection probabilities. However, this method would be the most computationally expensive method and would require additional work to write the algorithm and to code it into the combat simulation.

B. AUDITORY HAZARD ASSESSMENT ALGORITHM

1. Model Description

To remain consistent with the documentation for this program, the Auditory Hazard Assessment Algorithm is referenced to as AHAAH. [Ref 10] The AHAAH was created by the same researchers who wrote the ADM at the Army Research Laboratory. Both models were used to support MIL-STD-1474. The AHAAH "evaluates hazards for the human ear for sounds traveling toward the side of the head (a worst-case condition)." [Ref 10]

One of the outputs of the model is an array of sound pressure levels across the one-third octave bands. See Figures 37 and 38. The band outputs are a byproduct of the hazard analysis, which takes the waveform directly into a circuit model of the ear and calculates stresses in the inner ear using numerical integration of the equations of motion. The band spectrum results from a Fourier transformation of the waveform. The spectrum is used to measure hearing protector insertion losses and to simulate the pressures under a hearing protector. For this thesis, the AHAAH was used to provide sound signature for use with the ADM. [J. T. Kalb, personal communication, May 19, 2005]

Figure 37 shows the graphical output from the AHAAH for the single shot of an M16. The top graph displays the intensity of the sound across the one-third octave band or the signature of the sound. The ADM used this information. The bottom graph shows the pressure of the sound in reference to the time elapsed.

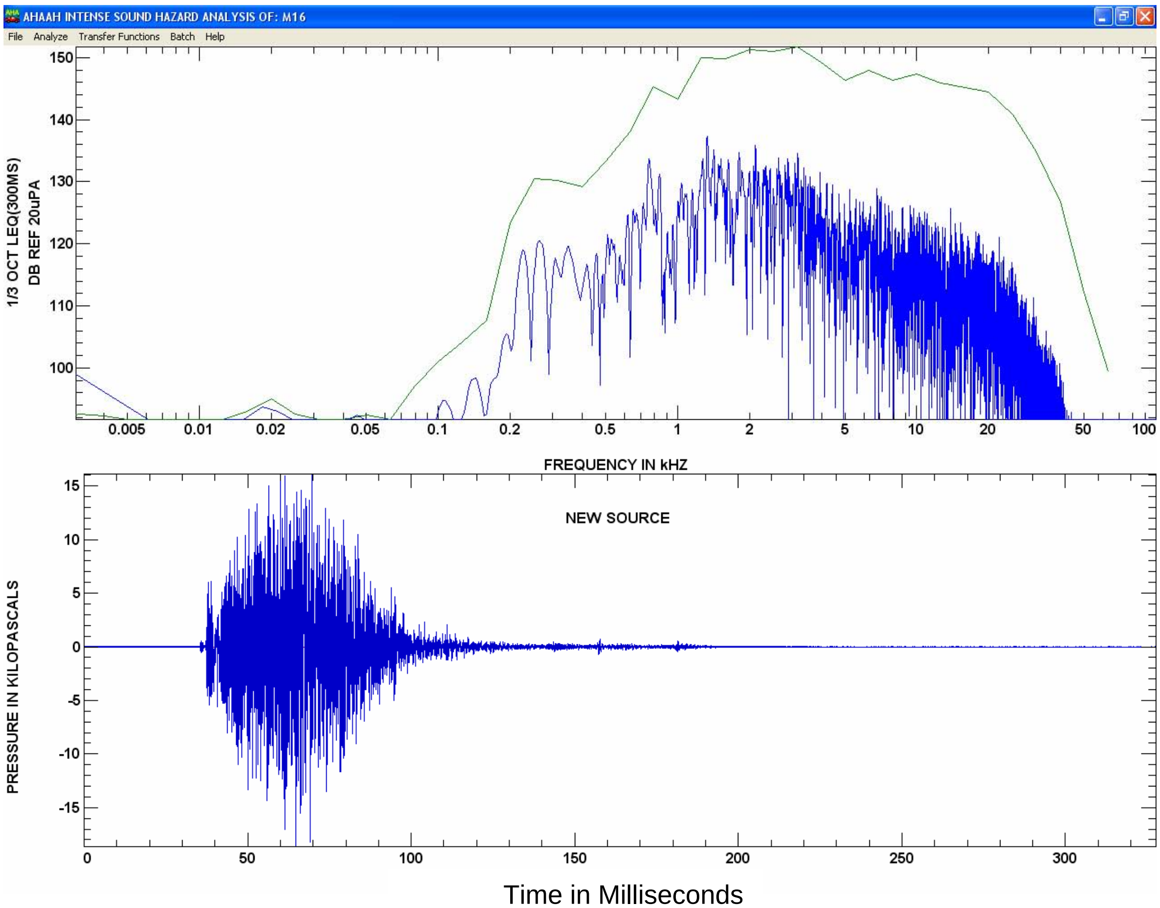

Figure 37. AHAAH Graphical Output for an M16 Burst

Figure 38 shows the AHAAH spreadsheet output for the single shot of an M16. The left column provides the center frequencies of the one-third octave band, and the right column provides the intensities in dB of the sound at the given frequency. The ADM uses the intensity values from the range 50 Hz to 10,000 Hz.

Leq300ms: steady 1/3 octave band sound pressure levels which when integrated for 300 milliseconds have energies equal to that of the signal.

Freq(Hz)	Leq(dB)
2.51	92.47
3.16	92.56
3.98	92.23
5.01	90.63
6.31	84.87
7.94	80.91
10.00	83.28
12.59	89.55
15.85	93.02
19.95	95.04
25.12	92.58
31.62	90.98
39.81	91.52
50.12	92.42
63.10	85.65
79.43	97.10
100.00	100.96
125.89	104.26
158.49	107.64
199.53	123.37
251.19	130.55
316.23	130.25
398.11	129.24
501.19	133.21
630.96	138.06
794.33	145.33
1,000.00	143.30
1,258.93	150.03
1,584.89	149.79
1,995.26	151.33
2,511.89	150.99
3,162.28	151.71
3,981.07	149.24
5,011.87	146.39
6,309.57	147.97
7,943.28	146.28
10,000.00	147.30
12,589.25	145.84
15,848.93	145.23
19,952.62	144.49
25,118.86	140.80
31,622.78	134.81
39,810.72	126.69
50,118.72	112.26
63,095.73	99.44

Figure 38. AHAAH Spreadsheet Output for an M16 Burst

2. Model Use

Dr. Ellen Haas of the Army Research Laboratory provided audio Compact Disks (CDs) containing approximately 170 sound recordings. Dr. Kalb converted these audio sounds into computer WAV files and also provided the AHAAH. The M16 and AK47 assault-rifle WAV files were selected, and the AHAAH was used to extract the sound signatures across the one-third octave band for these weapons.

C. COMBAT^{XXI}

1. Model Description

The Combined Arms Analysis Tool for the 21st Century (COMBAT^{XXI}) is a high-resolution, closed-form, stochastic, analytical combat simulation being developed by TRAC-WSMR and the Marine Corps Combat Development Command (MCCDC). COMBAT^{XXI} is a replacement for the Combined Arms and Task Force Evaluation Model (CASTFOREM) and is intended to support the analysis of force design, operational requirements, war-fighting experiments, and weapon-system development. This model combines hard-coded physical algorithms, primitive behaviors, and tactical behaviors with user-defined scenarios and rules to detail the interactions between the individual entities in the simulation. [Ref 22]

2. Model Implementation

The localization algorithms and data from the ADM and AHAAH were implemented with COMBAT^{XXI} in the following manner. The AHAAH was used to extract sound signatures for the AK-47 and M-16 assault-rifles from WAV file recordings. These sound signatures were input into the ADM, and the ADM was used to determine the ranges at which these sounds had a 99% probability of detection. These ranges were then coded into COMBAT^{XXI} as part of a cookie-cutter sound-detection routine.

In addition to the aforementioned work, several additional algorithms were hard-coded into COMBAT^{XXI} to make this concept feasible in the combat simulation: Sound signatures were mapped to the appropriate weapon systems, a trigger was created for a "make a noise" event, and a cookie-cutter method was implemented to cue the entities within the range of the sound to allow the entities to detect and to localize objects. The added code provides the framework that allows users to customize sound use for their scenarios.

In order to use the sound propagation, detection, and localization routine in COMBAT^{XXI}, a user must first include sound detection and localization as a behavior rule when creating the scenario. This is done by using the “Rule Library Builder” to assign a “Rule Template” to the unit template, units, or entities for which the user wants to have hearing capability. Within the “Select Trigger Events” section of the window, the user selects the “SoundHeard” option under the “detection” functionality directory. See Figure 39. Once the “SoundHeard” functionality is enabled, a user may either create a rule or import an existing rule for use with all the sounds replicated in the simulation. Currently, COMBAT^{XXI} models sounds for the M-16 and AK-47 assault-rifles. However, the code writers have the option of adding more sounds by using the AHAAH and ADM, as described in previous sections.

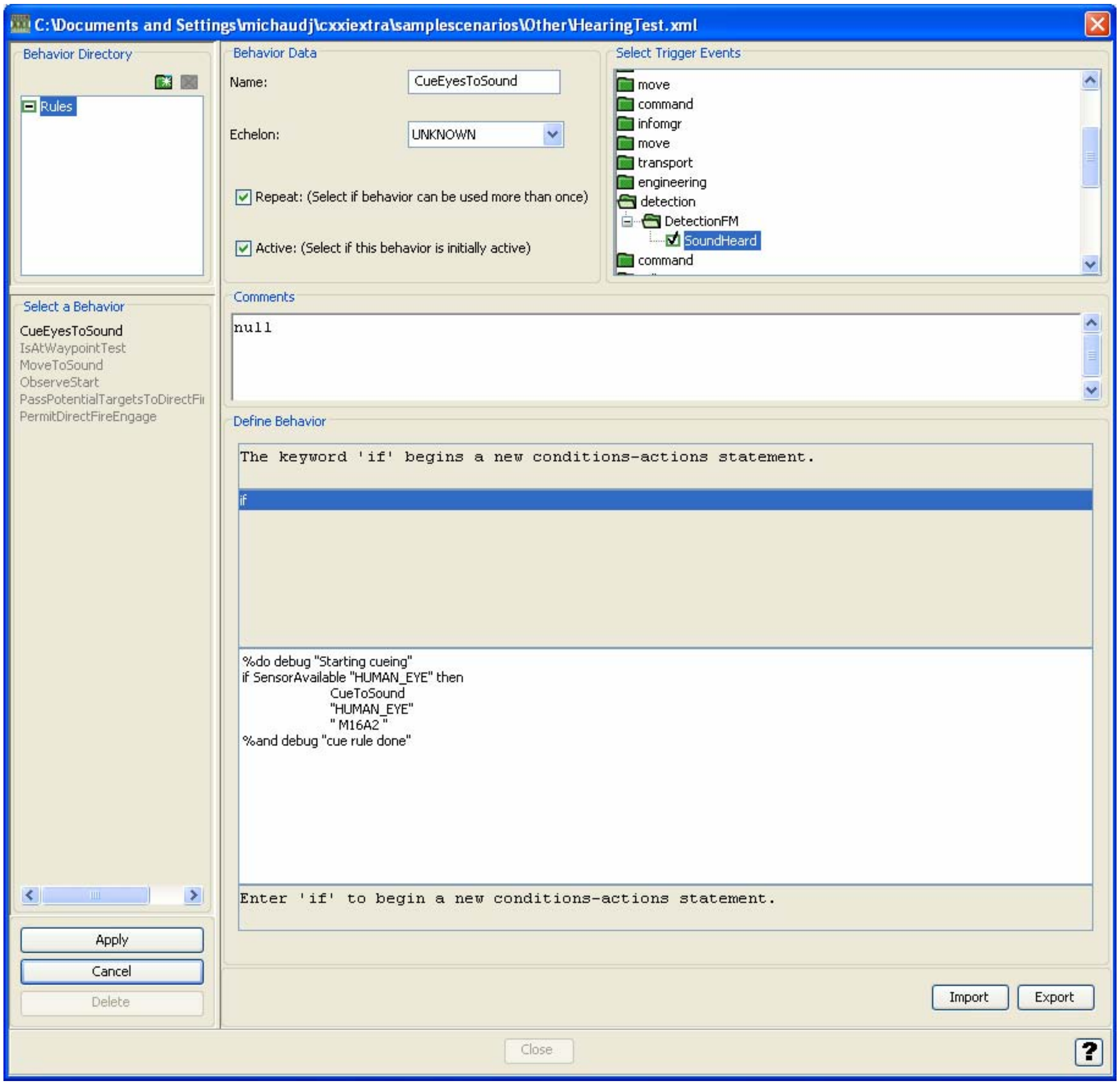

Figure 39. Rule Library Builder

To create a rule, the user names and defines his custom behavior in the "Rule Data" and "Define Rule" sections of the window. Figure 39 shows the custom behavior rule, "CueEyesToSound." In the "Define Behavior" block of Figure 39, the rule directs the selected entities to cue their eyes to all sounds in the simulation, except for the M-16 assault-rifle. The exception for the M-16 was created to prevent the entities from cuing to members of their own unit.

To import a rule, the user simply clicks on "Import" at the bottom of the window, selects the "Detection Folder," then selects the desired rule. See Figure

40. Once the rule is imported, the user may alter the rule by changing the commands in the "Define Behavior" box. The rules currently available for import with the "SoundHeard" functionality are "CueEyesToSound" and "MoveToSound". The "CueEyesToSound" rule is described above. The "MoveToSound" rule directs an entity to move toward a sound for a specified period of time. If a target is not found within that time period, the entity continues with the mission that it was performing prior to hearing the sound.

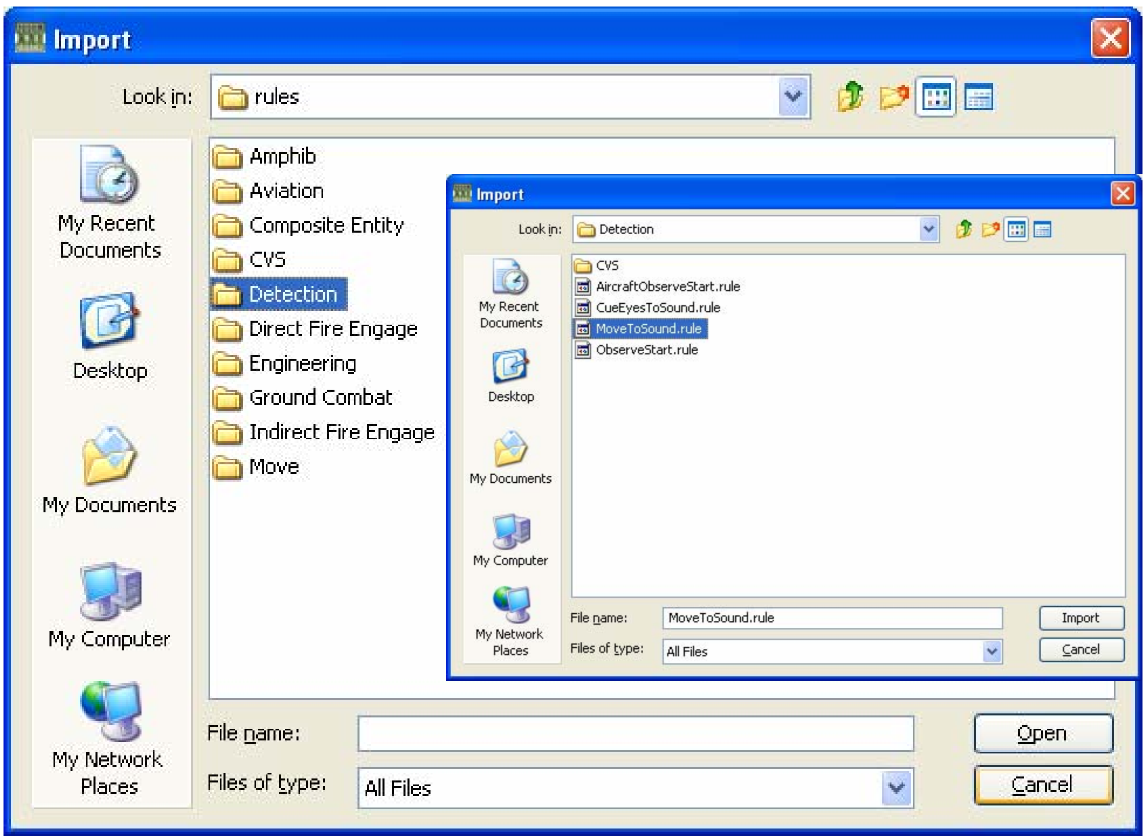

Figure 40. Import a Behavior Rule

3. Application

By adding the sound-cuing capability to COMBAT^{XXI}, users can specify how entities react to sounds in the simulation in any manner they choose. As an example, a user may instruct entities to assume a protective posture immediately when they hear weapon fire. This flexibility gives analysts the ability to examine the effects of human hearing and acoustic detection in many different ways during combat operations. Possible analysis includes a Soldier’s ability to survive attacks by assuming a protective posture when reacting to indirect or direct-fire weapons, the effects of quieter equipment on nondetectability, a Soldier’s ability to locate targets in poor visibility situations (smoke, heavy vegetation, low light, enemy positioned behind walls, etc), a Soldier’s ability to gather intelligence, or the effects of non-lethal acoustic weapons.

VI. EXPERIMENTAL DESIGN AND DATA ANALYSIS

A. EXPERMIMETNAL DESIGN

1. Scenarios

This experiment was designed with two scenarios to determine whether the addition of the ADM, AHAAH, and sound-localization algorithms to COMBAT^{XXI} causes computer-generated entities to act more realistically than they would without the additions. The first scenario was designed to detect the differences between COMBAT^{XXI} and COMBAT^{XXI} with the auditory-detection capabilities added. The second scenario was designed to detect the differences between COMBAT^{XXI} with perfect sound-localization capabilities and COMBAT^{XXI} with imperfect sound-localization capabilities. Perfect sound localization provided computer-generated entities with exact location information for a sound source. Imperfect sound localization used the sound-localization algorithms developed in this thesis to provide an entity with inaccurate location information for a sound source.

a. Scenario A

This scenario was designed to detect differences between COMBAT^{XXI} and COMBAT^{XXI} with the auditory-detection capabilities added. The test scenario consisted of one red squad comprised of 14 members and one blue squad comprised of ten members. The red squad's mission was to kill as many entities on the blue team as possible, and the blue team's mission was to move along a route from an assembly area to a final objective. The terrain was rolling hills with mixed vegetation (Folda Gap, Germany). The members of the blue team moved in a column formation with staggered orientation (i.e., the first person faced forward, the second faced to the right, the third to the left, the fourth to the right, and so forth, with the final person faced rearward). The red team was dispersed across the terrain with one ambush set up along the blue route. Only the blue team had the ability to “hear.”

Figure 41 graphically depicts Scenario “A.” The circles represent individual Soldiers. The lines on the circles point in the directions in which the

Soldiers are currently oriented. The blue team is organized in a row at the bottom left corner of the figure. The solid line that runs from the lower left to the upper right of the figure is the path traveled by the blue team. Members of the red team are dispersed along the blue team's path. The shaded areas represent terrain consisting of scrub brush.

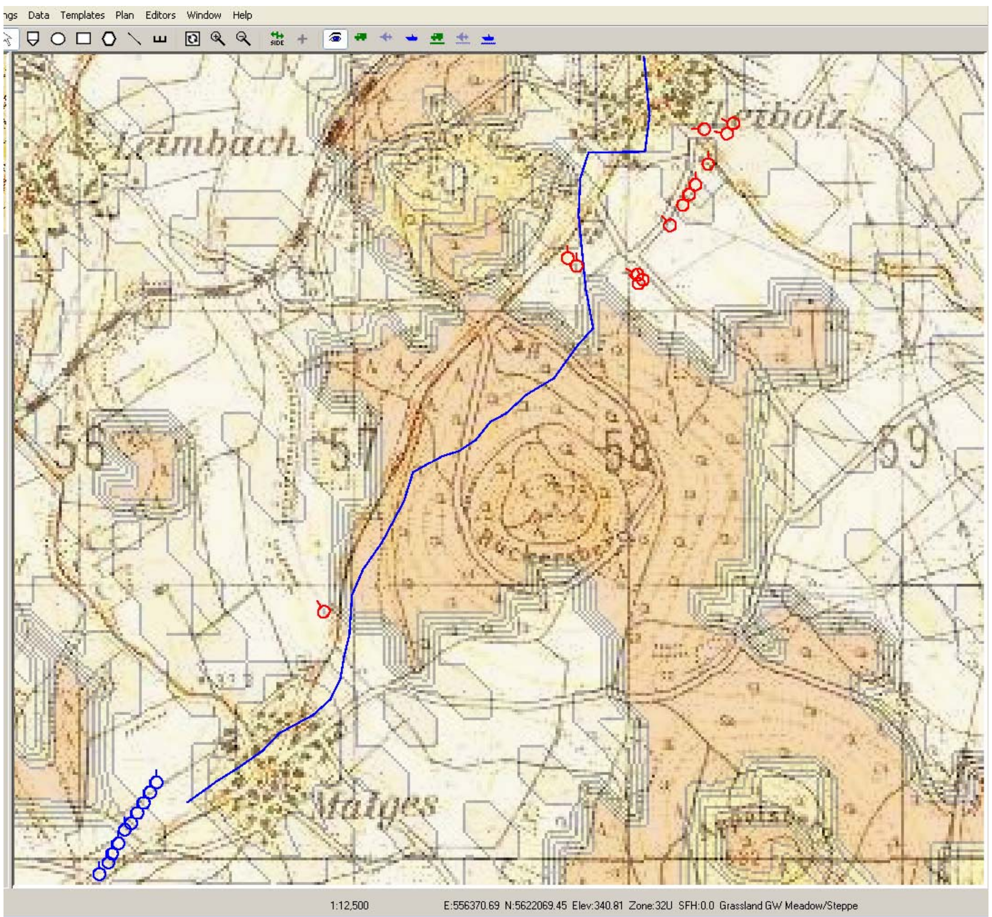

Figure 41. Scenario A

b. Scenario B

This scenario was designed to detect the difference between COMBAT^{XXI} with perfect sound-localization capabilities and COMBAT^{XXI} with imperfect sound-localization capabilities. This scenario consisted of twelve blue entities divided equally into four squads and two individual red entities. The missions of the red and blue teams were the same as in Scenario "A."

The two red entities were located in hide positions on opposite sides of the blue team's path for an ambush. The positioning of the red entities was intended to test the accuracy of the blue entities' cuing abilities. With Scenario "A," 14 red entities were dispersed along the blue team's path. The large number of entities and dispersion provided a "target rich" environment for the blue team. In Scenario "B," the scarcity of targets, as well as their hidden positions, should have made it more difficult for the blue entities to acquire them.

The blue squads traveled down the path, separated by a fixed time interval. The small number of blue entities in each squad and the time separation between the squads were intended to mitigate the effects of "massed cuing" as demonstrated in Scenario "A." "Massed cuing" is the effect of multiple entities simultaneously cuing to a target and at least one of them acquiring the target. With a large number of entities, as in Scenario "A," there is a much higher probability of this happening.

Figure 42 graphically represents Scenario "B." As with Scenario "A," the circles represent individual Soldiers. The lines on the circles point in the directions in which the Soldiers are currently oriented. The blue team is located at the bottom of the figure and is organized in four teams of three entities. The blue line that runs from the bottom to the top of the figure is the path traveled by the blue team. Two red entities are located on opposite sides of the blue team's path in the top third of the figure. The shaded areas represent terrain consisting of scrub brush.

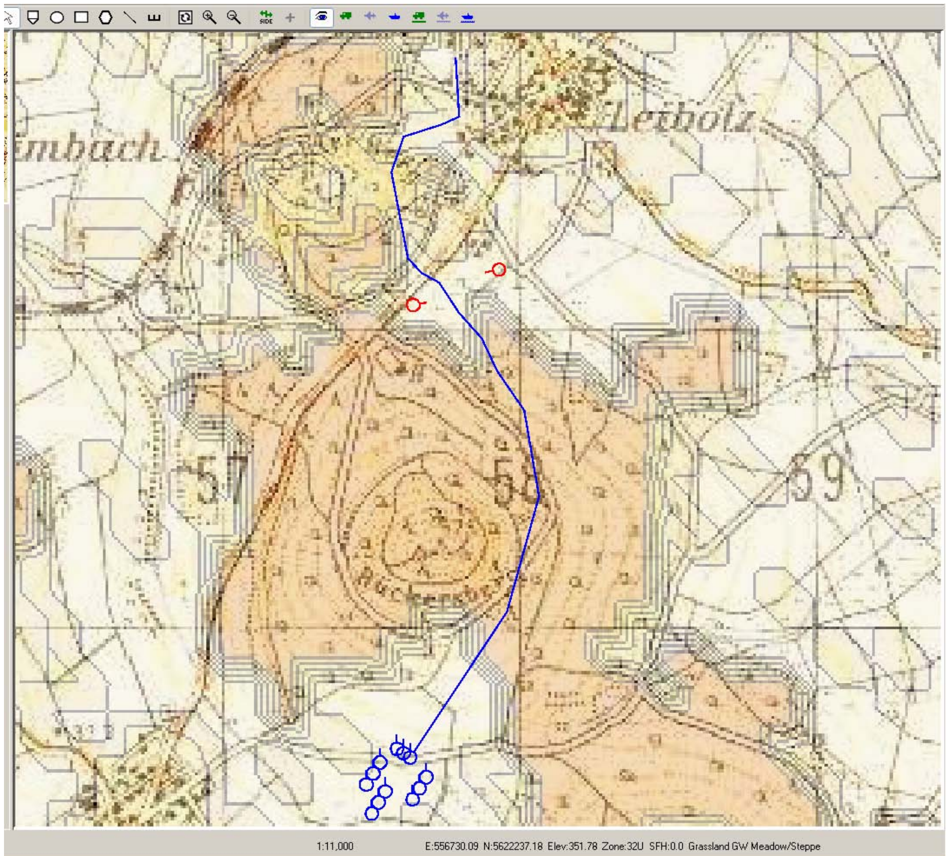

Figure 42. Scenario B

2. Cases

Each scenario was executed using five cases: "No Sound" (NS), "Sound with No Errors While Moving" (SNM), and "Sound with No Errors While Pausing" (SNP), "Sound with Errors While Moving" (SEM), and "Sound with Errors While Pausing" (SEP). The events "Errors" and "No Errors" were included to test whether or not the sound-localization algorithms that were added to COMBAT^{XXI} caused a difference in the outcome of the simulation. The events "Moving" and "Pausing" were added to the scenarios because it was thought that the algorithm for vision might yield different acquisition results for entities that are moving as opposed to stationary.

The first case, NS, was the base case with COMBAT^{XXI} under its current configuration. The case, SNM, was COMBAT^{XXI} with the added auditory-detection capabilities of the ADM and AHAAH and using the "CueEyesToSound" rule, as described in the previous section. This case provided entities with perfect target location while they moved constantly toward their objective, even when a sound was heard. The SNP case was similar to SNM, except the entities briefly stopped their movement when a sound was heard in order to attempt to better locate the source of the sound. The final two cases, SEM and SEP, were similar to SNM and SNP with the difference that SEM and SEP provided entities with target location error. In all five cases, the entities could fire only upon enemies that were acquired via vision.

For case NS, the entities could acquire their targets through vision if their eyes were already oriented in the direction of the target and the target fell within the entities' field of view. Entities did not reorient their eyes after shots were fired. This is the way COMBAT^{XXI} is currently configured. For the other cases, entities initially acquired targets through vision if they were oriented in the direction of the target, as was done with NS. However, once a shot was fired by a red entity, each blue entity heard the shot and reoriented its eyes in the direction from which the shot originated. For SNM and SNP, the entities looked directly at the target, and for SEM and SEP, the entities looked in the general direction of the target. The entities then used their vision to attempt to locate the target. If the target was located, the entities engaged the target.

3. Measures of Effectiveness (MOE)

The Measure of Effectiveness (MOE) used for both scenarios was the number of fatal wounds inflicted on blue entities. This MOE defines the accomplishment of both the red and blue squads' missions. The red squad's mission was to kill as many blue as possible. The blue squad's mission was to move on to a final objective. Any blue entity not killed eventually moved to the final objective. A low number of blue fatal wounds indicated the blue entities were able to adequately defend themselves, resulting in more blue team members reaching their final objective.

B. DATA ANALYSIS

This analysis was conducted to determine whether there were differences among COMBAT^{XXI} as currently configured, COMBAT^{XXI} with auditory-detection capabilities and perfect localization added, and COMBAT^{XXI} with auditory-detection capabilities and imperfect localization added. The differences were then examined to determine whether the combination of the sound-localization algorithms, ADM, AHAAH, and COMBAT^{XXI} more realistically represented the individual Soldier.

1. Overview

S-PLUS statistical software was used to conduct the data analysis. One hundred trials of each case were run for each scenario. First, the data for each of the five cases was compared against every other case within each scenario. Next, the case "Sound with Errors While Moving" was combined with the case "Sound with Errors While Pausing" to form the group "Errors" and the case "Sound with No Errors While Moving" was combined with the case "Sound with No Errors While Pausing" to form the group "No Errors." These two groups were then compared within each scenario. Finally, the data for both scenarios were combined in a similar manner, as just described, to create one large data set with "Errors" and one with "No Errors." Shapiro-Wilks Tests for Normality, Friedman's Tests for a Randomized Block Experiment, Wilcoxon Rank-Sum Intervals utilizing the Bonferroni Correction, descriptive statistics, and boxplots were used to analyze the data produced by these trials. Finally, "face validation" was used to determine which implementation of COMBAT^{XXI} provided the most realistic results.

2. Data Collected

The data collected for each trial was the number of fatal wounds inflicted on the blue team. Each case was run 100 times. This number was selected because COMBAT^{XXI} currently has the ability to use 100 random number seeds. Therefore, each case was run for each scenario once with each random number seed. See Appendix B for data.

3. Shapiro-Wilks Test for Normality

Shapiro-Wilks Tests for Normality were conducted to determine whether the data produced for each scenario by each case in this experiment could have come from the normal distribution. The null hypothesis was that the distribution producing the data was not significantly different from the normal distribution. The alternative hypothesis was that the distribution producing the data was significantly different from the normal distribution. The results of the Shapiro-Wilks Tests performed on this data set showed that the data produced by each of the five cases under both scenarios, as well as the combined groups “Errors” and “No Errors,” were significantly different from the normal distribution. This result led to the selection of distribution-free hypothesis tests and intervals to analyze this data set further. [Ref 6]

4. Friedman’s Test for a Randomized Block Experiment

Friedman’s Test for a Randomized Block Experiment is a distribution-free hypothesis test that is used to compare the expected values of multiple (more than two) samples. This test is often used in place of ANOVA when data are not normally distributed. [Ref 6] For this experiment, Friedman’s Test was used to determine whether any differences existed among the expected values of the data for the five various cases within each scenario. This test was not used to compare the combined samples of “Errors” and “No Errors.”

For this data analysis, the treatments were the various cases while the blocks were the observation number for each data point. The selection of the cases as treatments is relatively straight forward and does not warrant further discussion. However, the selection of observation number as the selection for blocks does. Each observation for each trial was gained by using a specific random number seed. Therefore, each observation within a case used the same seed as the same observation number within all the other cases. For example, observation “1” for case NS used the same random number seed as observation “1” for case SNM, as well as observation “1” for case SEM. This resulted in 100 blocks used for the Friedman’s test, one for each of the 100 random number seeds.

The Friedman’s Test conducted on the data for each of the five cases showed that differences existed among the cases within Scenario “A” as well as among the cases within Scenario “B.” This test, however, did not provide any information about which cases were different from one another. In order to do this, the Wilcoxon Rank-Sum Interval using the Bonferroni Correction was used.

5. Wilcoxon Rank-Sum Interval Using the Bonferroni Correction

The Wilcoxon Rank-Sum Interval is a distribution-free hypothesis test that compares the locations of two samples to determine if they are different from each other. This test is often used in place of the t-test or z-test when data are not normally distributed. For this analysis, the paired Wilcoxon Rank-Sum Interval was used to capture the effects of using the same random number seed for each observation with each case. [Ref 6]

The Bonferroni Correction is an adjustment applied to a family of multiple hypothesis tests to control the family-wise error rate. The Bonferroni Correction is applied by dividing the alpha value by the number of comparisons performed. This method keeps the probability of rejecting at least one null hypothesis when it is true (Type I error) below the desired alpha value. [Ref 6]

a. Comparison of Five Cases within Each Scenario

For this situation, five cases were compared against one another within each scenario. Therefore, a total of ten Wilcoxon Rank-Sum Tests were conducted and the alpha value was divided by ten before comparing it to the *p*-values. Tables 4 and 5 provide the *p*-values that resulted from the paired Wilcoxon Rank-Sum Intervals using the Bonferroni Correction.

Scenario A					
	NS	**SEM**	**SEP**	**SNM**	**SNP**
NS	**X**	0	0	0	0
SEM	0	**X**	0.1277	0.0648	0.9194
SEP	0	0.1277	**X**	0.0003	0.2076
SNM	0	0.0648	0.0003	**X**	0.2179
SNP	0	0.9194	0.2076	0.2179	**X**

Table 4. *p*-values for Scenario “A”

Scenario B					
	NS	**SEM**	**SEP**	**SNM**	**SNP**
NS	**X**	0	0	0	0
SEM	0	**X**	0.3303	0.1237	0.2541
SEP	0	0.3303	**X**	0.0027	0.0087
SNM	0	0.1237	0.0027	**X**	0.7289
SNP	0	0.2541	0.0087	0.7289	**X**

Table 5. *p*-values for Scenario "B"

For both scenarios using an alpha of 0.05 (0.005 once the Bonferroni Correction was applied), the case NS was significantly different from all other the cases, and cases SEP and SNM were different from each other. For Scenario "B," the case SEP was significantly different from both SNM and SNP for when an alpha of 0.1 was used.

b. Comparison of "Errors" and "No Errors" within Each Scenario

Within each scenario, the sample "Errors" was compared to "No Errors" using the Wilcoxon Rank-Sum Test. This test resulted in *p*-values of 0.0289 for Scenario "A" and 0.0029 for Scenario "B." This test demonstrates that there are significant differences existed between these two data samples.

c. Comparison of "Errors" and "No Errors" Combining Both Scenarios

The sample titled "Errors" from Scenario "A" was combined with the sample of the same name from Scenario "B" and compared against the combined sample titled "No Errors." The Wilcoxon Rank-Sum Test conducted on these two samples yielded a *p*-value of 0.0003 which demonstrates again that "Errors" is significantly different from "No Errors."

6. Descriptive Statistics and Boxplots

a. Comparison of Five Cases within Each Scenario

1. Scenario "A": The descriptive statistics shown in Table 6 and boxplots in Figure 43 are arranged in order of decreasing means. The case with the largest mean is on the left, progressing to the case with the lowest mean on the right. A high mean represents poor performance by the blue team. Note that the NS case was both significantly different from all the other cases and resulted in the worst blue team performance, regardless of the statistics used for comparison. The SEP case yielded poorer blue team performance than did the SNM case when using the max or mean statistics for comparison and was similar when using the median, mode, or min statistics. The analysis from the previous section proved SEP to be significantly different from SNM. The boxplots in Figure 43 reinforce these results graphically. It is interesting to note that the cases that include errors appear to be grouped in a manner that suggests they perform slightly worse than the cases that do not include errors. This observation agrees with the analysis above that determined there is a difference between cases with “Errors” and cases with “No Errors.”

Descriptive Statistics for Scenario A

	NS	SEP	SEM	SNP	SNM
Size	100	100	100	100	100
Max	12	6	9	5	4
3rd Quartile	10	2	2	2	2
Median	10	1	1	1	1
Mean	9.66	1.59	1.4	1.35	1.08
Mode	10	1	0	1	1
1st Quartile	9	1	0	0	0
Min	4	0	0	0	0
Std Dev	1.26	1.22	1.41	1.26	0.94

Table 6. Descriptive Statistics for Scenario “A”

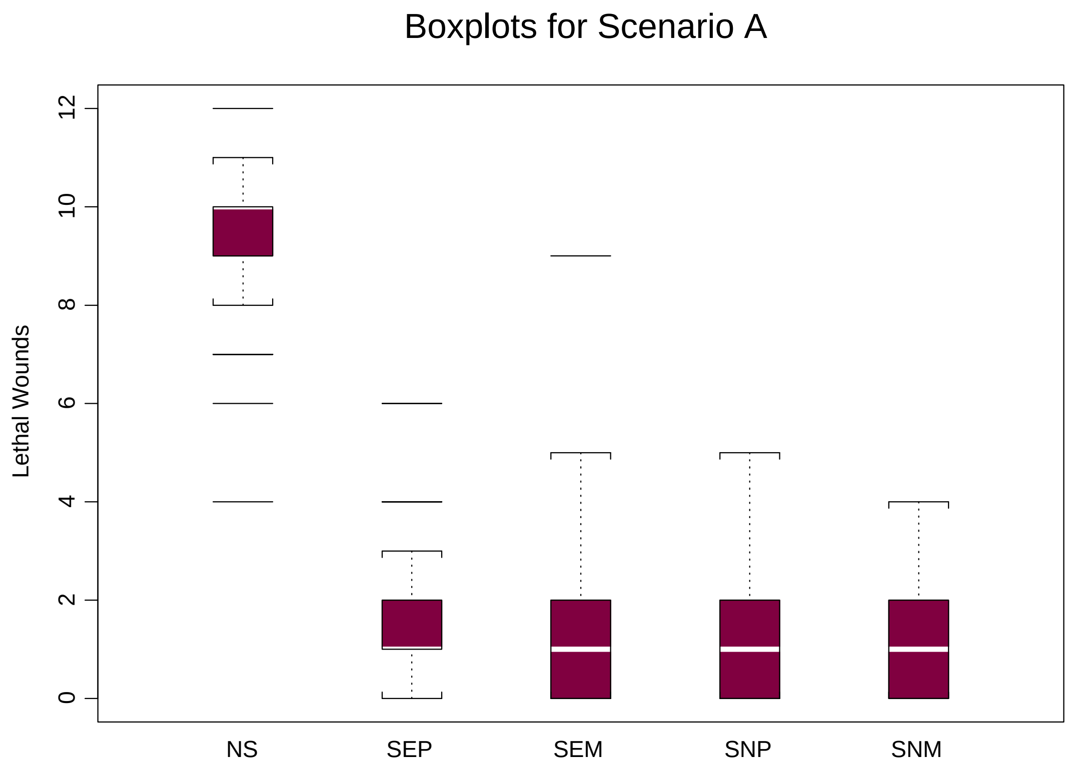

Figure 43. Boxplots for Scenario A

2. Scenario “B”: As with Scenario “A”, the descriptive statistics in Table 7 and boxplots in Figure 44 are arranged from left to right, in order of decreasing means. Again, NS was significantly different from all the other cases and resulted in the worst blue team performance regardless of the statistics used for comparison. The case SEP yielded poorer blue team performance than did the SNM and SNP cases when using the max or mean statistics for comparison and was similar when using the median, mode, or min statistics. The analysis from the previous section proved SEP to be significantly different from SNM when using an alpha of 0.05 and different from SNM and SNP when using an alpha of 0.01. The boxplots in Figure 44 reinforce these results graphically. As with Scenario “A,” the cases with errors appeared to perform slightly worse than the cases without errors.

Descriptive Statistics for Scenario B

	NS	SEP	SEM	SNP	SNM
Size	100	100	100	100	100
Max	13	11	7	6	7
3rd Quartile	12	2.25	2	1	1
Median	12	1	1	1	1
Mean	10.89	1.6	1.25	1.09	0.97
Mode	12	0	0	0	0
1st Quartile	11.75	0	0	0	0
Min	1	0	0	0	0
Std Dev	2.49	1.94	1.44	1.40	1.34

Table 7. Descriptive Statistics for Scenario B

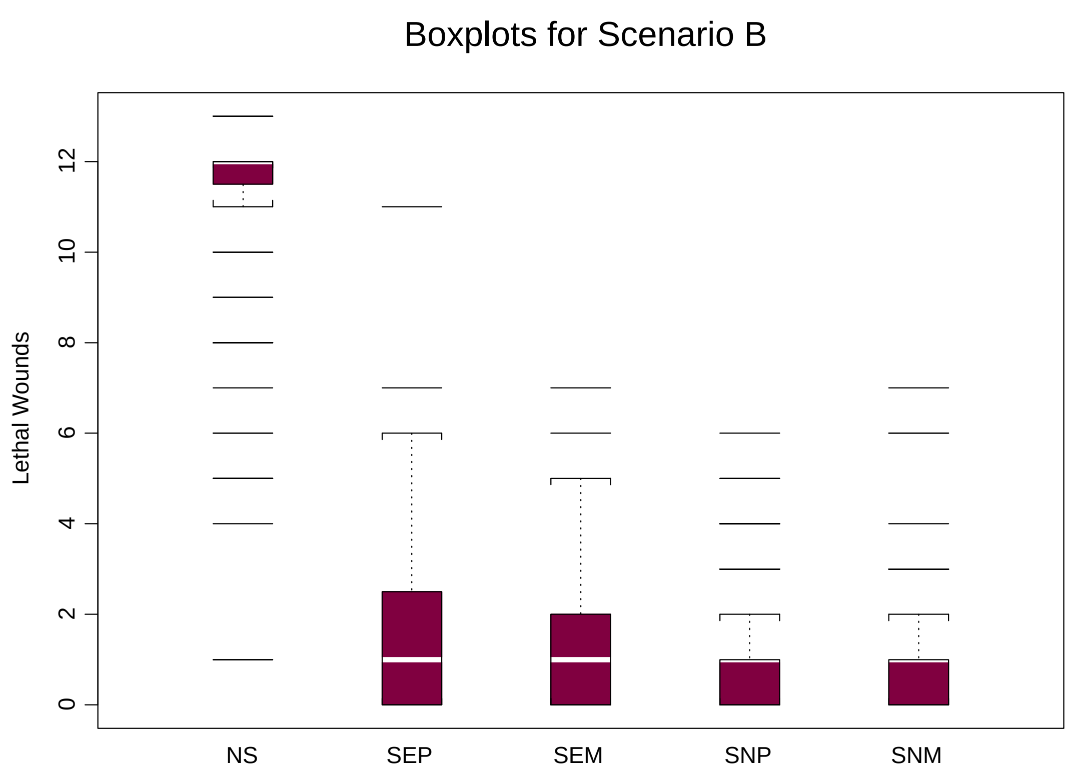

Figure 44. Boxplots for Scenario B

b. Comparison of "Errors" and "No Errors" within Each Scenario

Combining the cases with "Errors" into one group and the cases with "No Errors" into another group provides information on how the localization algorithm created in this thesis performs when the entities either pause or continue moving while localizing sounds. In this case each group contains performance information for entities that continue to move while localizing a sound source, as well as for entities that stop to better pinpoint the source of a sound. As before, the cases are ordered with the highest mean on the left and lowest mean on the right within each scenario. Table 8, Figure 45, and Figure 46 indicate that the case with "Errors" results in slightly worse performance for the blue team in both scenarios than the case with "No Errors."

Descriptive Statistics for "Errors" and "No Errors"
within Each Scenario

	Scenario A		Scenario B	
	Error	No Error	Error	No Error
Size	200	200	200	200
Max	9	5	11	7
3rd Quartile	2	2	2	1
Median	1	1	1	1
Mean	1.50	1.22	1.43	1.03
Mode	1	1	0	0
1st Quartile	1	0	0	0
Min	0	0	0	0
Std Dev	1.32	1.12	1.71	1.37

Table 8. Descriptive Statistics for "Errors" and "No Errors" for Each Scenario

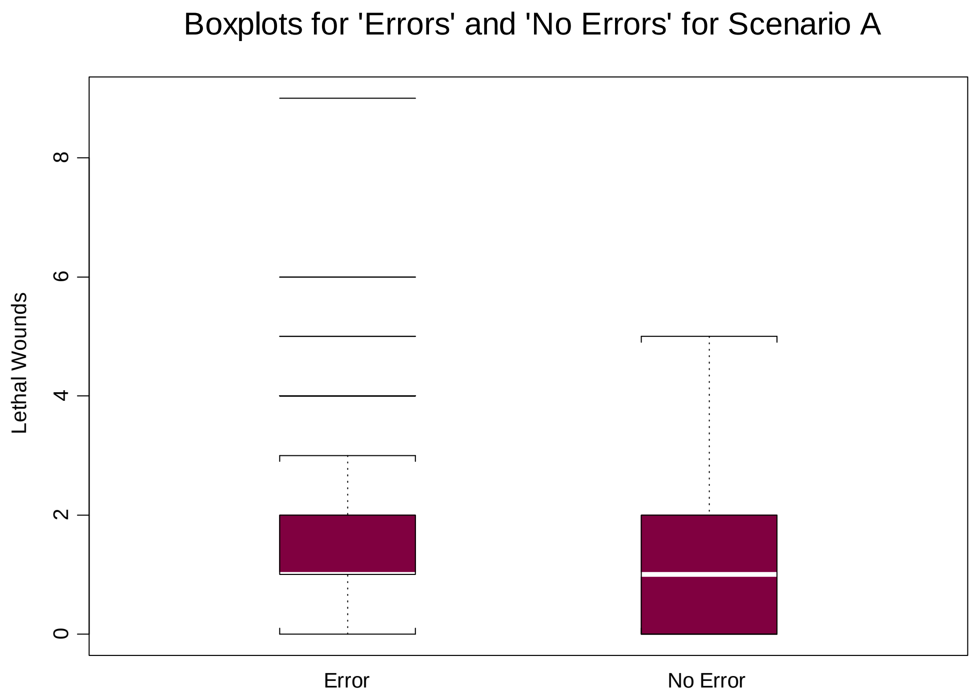

Figure 45. Boxplots for “Errors” and “No Errors” for Scenario “A”

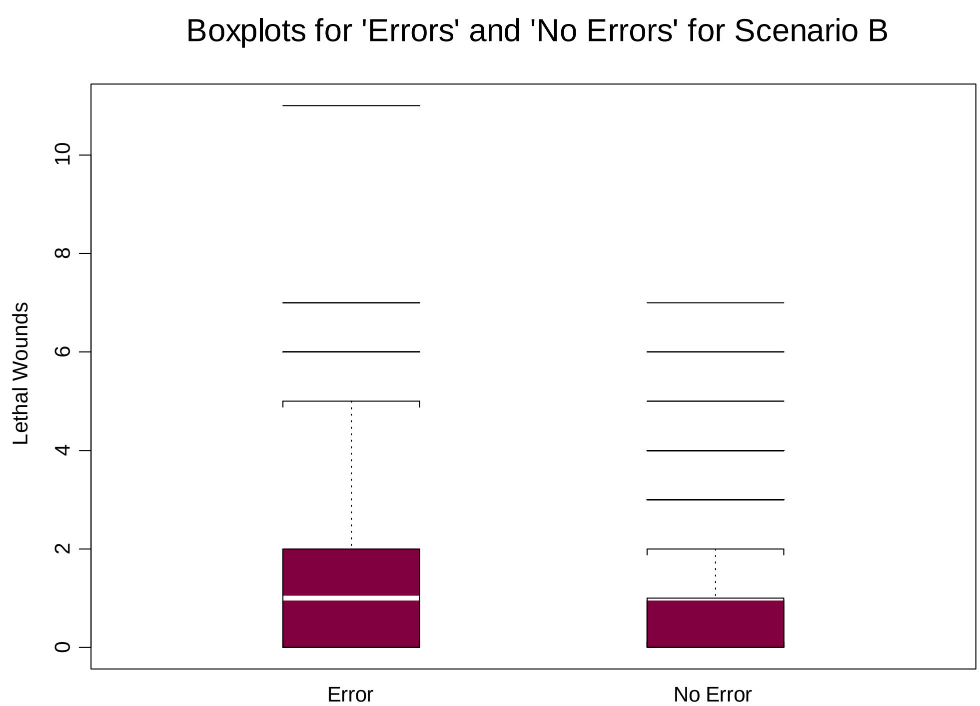

Figure 46. Boxplots for “Errors” and “No Errors” for Scenario “B”

c. *Comparison of "Errors" and "No Errors" Combining Both Scenarios*

Combining the two conglomerated groups from both scenarios into two large groups with "Errors" and "No Errors" provides information on how the localization algorithm performs with entities acting in various manners and in multiple situations. The data in Table 9 and Figure 47 are arranged as before with the best performing case on the right and the worst performing on the left. This table and figure again demonstrate that the case with "Errors" provides slightly worse performance than the case with "No Errors."

Descriptive Statistics for "Errors" and "No Errors"

	Error	No Error
Size	400	400
Max	11	7
3rd Quartile	2	2
Median	1	1
Mean	1.46	1.12
Mode	1	0
1st Quartile	0	0
Min	0	0
Std Dev	1.53	1.25

Table 9. Descriptive Statistics for "Errors" and "No Errors"

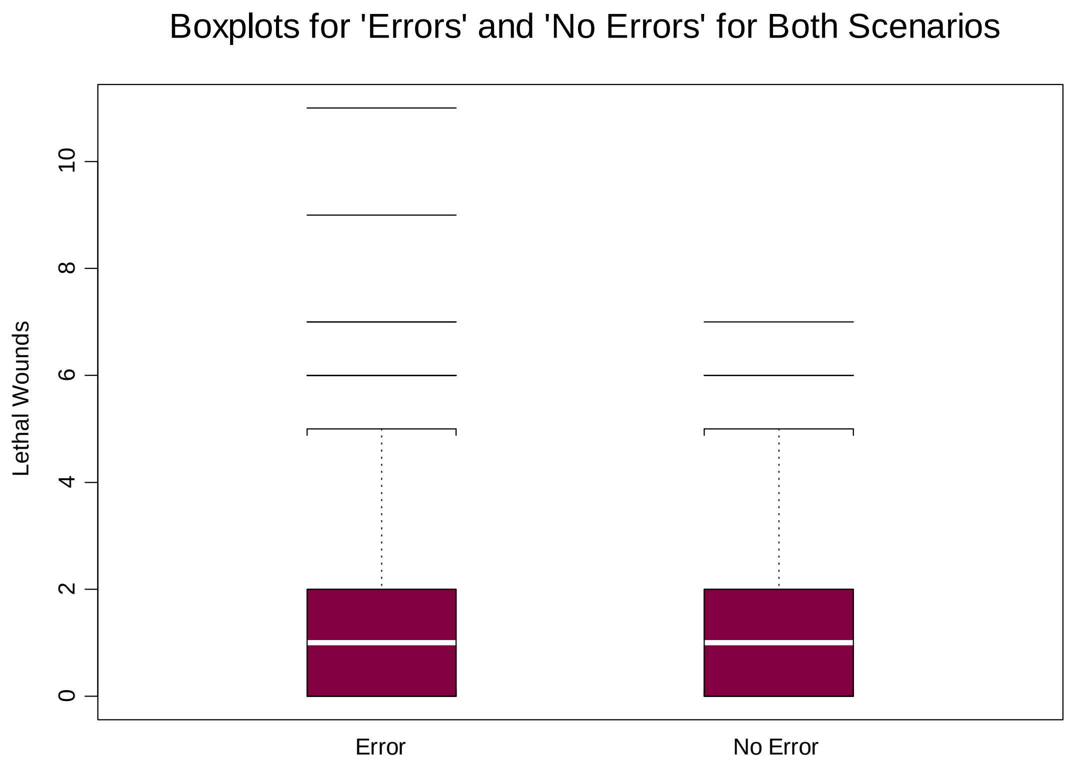

Figure 47. Boxplots for “Errors” and “No Errors” for Both Scenarios

7. Face Validation

Face validation is an informal validation technique in which "project team members, potential users of the model, and subject matter experts (SMEs) review simulation output (e.g., numerical results, animations, etc.) for reasonableness. They use their estimates and intuition to compare model and system behaviors subjectively under identical input conditions and judge whether the model and its results are reasonable." [Ref 24]

From the results of the data analysis, clearly, the cases using the auditory detection capability in the simulation produced outcomes that were statistically different and much more favorable for the blue team than without this capability. By observing how the entities reacted while the simulation was running, one sees that the entities behaved in a more realistic manner with auditory detection included than without. Without auditory detection in the simulation, the entities did not react to an attack unless they were attacked directly from the front. With the auditory detection, entities reacted whenever a shot was fired, even if the shot was not directed at them.

The data analysis also demonstrated that COMBAT^{XXI} using localization errors produced results that were statistically different and slightly less favorable than without the errors applied. The only exception to this was when the five separate cases were compared against each other. In this instance, some of the cases with errors applied were different from some of the cases without errors, but not all. This leads to the conclusion that the addition of sound-localization errors to the simulation may produce outcomes slightly less favorable for the entities that use the errors.

Additionally, it follows conventional wisdom that applying errors to a localization routine would result in fewer target acquisitions, therefore fewer red kills, and ultimately more blue casualties than when providing entities with perfect information. This leads to the conclusion that using the sound-localization algorithm developed in this thesis provides outcomes that are consistent with conventional logic.

The face validation method was used to validate the use of the ADM, AHAAH, and sound-localization algorithm for use with COMBAT^{XXI}. It was evident that the addition of the auditory-detection capability and sound-localization algorithms to COMBAT^{XXI} caused the entities to act in a manner that was reasonable.

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VII. SUMMARY, FUTURE RESEARCH, AND CONCLUSIONS

A. SUMMARY

The focus of this thesis was to enhance combat simulations by providing a method by which computer-generated entities could use their sense of hearing to detect and to locate objects through a phenomenon known as "sound localization." This goal was accomplished by providing three tools: a program that models sound propagation and auditory detection, a program that extracts sound signatures from recordings, and algorithms that replicate a human's sound-localization abilities.

Examination of previous work showed that sound cuing in combat simulations, such as OneSAF, JANUS, and UCCATS, has primarily used simple sound propagation and auditory-detection models and that sound-localization studies have focused on replicating auditory cues with headphones to motivate a human to a desired response. Sound-localization studies demonstrated that a human's directional location error depends on the location of the sound source in relation to the head. A human's most accurate localization abilities are for sounds originating from the front of the subject near the horizon, and the worst performance occurs for sounds originating from behind the subject at a high or low elevation.

Studies of the phenomena of human auditory detection revealed that auditory detection is a function of both the physical characteristics of a sound (frequency, amplitude, temporal characteristics, and location) and a human's hearing threshold.

An analysis of sound-localization data, provided by the Air Force Research Lab, led to the creation of algorithms that realistically replicated a human's sound-localization abilities in a computer model. The results of this analysis are consistent with findings from previous studies.

Existing models were selected for use in this thesis. The ADM models both sound propagation and human auditory detection and accounts for many of the sound and human characteristics required for auditory detection. The AHAAH augments the ADM by providing sound signatures for any sound recorded in a WAV file format. COMBAT^{XXI} provides the computer-generated environment, entities, and behaviors to test the sound-localization methods developed in this thesis.

The sound-localization algorithms and data from the ADM and AHAAH were implemented in COMBAT^{XXI}. Scenarios were developed and data was analyzed to determine whether the sound-localization capabilities added to COMBAT^{XXI} made the entities in the simulation perform in a more realistic manner. The conclusion drawn from this analysis was that the addition of the auditory-detection and sound-localization capabilities to COMBAT^{XXI} did, in fact, cause the entities to behave more lifelike than without these capabilities added.

The final products of this thesis are a program that robustly models sound propagation and auditory detection, a program that extracts a sound signature for any sound recorded in a WAV file format, and algorithms that replicate a human's sound-localization abilities. These products enhanced the realism of COMBAT^{XXI} and might be configured for use in any combat simulation.

B. RECOMMENDATIONS FOR FUTURE RESEARCH

There are many possible areas for future work. Improvements may be made to the existing code, data for use with the current implementation of the algorithms may be collected, or new areas in which to use the models may be researched.

1. Improve Code

a. Convert Code into Java

The ADM has not been successfully converted from the Pascal computer language into the Java computer language. [Ref 13] Java is the language in which COMBAT^{XXI} is written. Translating the program into Java will allow one to directly calculate the probability of auditory detection given a distance from the sound source, thereby enhancing the stochastic modeling of sound localization. A Java version of the ADM will also allow a comparison of the three methods discussed in Chapter V, Section A for accuracy and efficiency. Additionally, the code may be modified during translation to provide more efficient computational performance.

b. Verification, Validation, and Accreditation

The ADM has not undergone official Verification, Validation, and Accreditation (VV&A). The AHAAH is currently undergoing this process. These programs and the localization algorithm should undergo a more robust VV&A than the face validation method used during this thesis.

2. Data Collection

a. Localization under Realistic Conditions

Sound-localization experiments conducted under more realistic conditions may provide data that improve localization algorithm performance. Trials may be administered in the presence of background noises instead of in a quiet room; with sound recordings rather than a signal burst that has a pure tone and fixed intensity; and while the subject is wearing equipment such as a helmet, earplugs, or communications headset.

b. Sound Library

Analysis performed to identify important sound cues in a combat environment will contribute to the creation of a sound library. The library should include recordings and information about the conditions at the time of recording that are compatible with the ADM and AHAAH. The COMBAT^{XXI} code writers already have access to two CDs of sounds, each with more than 80 sound effects, to support this work. The only drawback with these sounds is that the researcher would have to make and justify assumptions about the conditions at the time of the sounds' recordings.

3. New Research

a. Distance Estimation

Find and analyze data from sound localization experiments regarding distance estimation and create an algorithm that models distance estimation stochastically. This algorithm could be used to enhance the "MoveToSound" rule in the COMBAT^{XXI} "Rule Library" by providing an additional trigger to end this event with distance traveled. The rule would instruct an entity to stop moving toward the sound and to continue the previous mission once the designated time limit expired or once the entity traveled the distance it estimated to the sound source. Currently, the "MoveToSound" rule terminates only after the designated time has elapsed.

b. Effects of Signal Frequency and Intensity

Research the effects of signal frequency and intensity on sound localization. Write a localization algorithm that accounts for sound intensity and frequency propagated at the listener's location.

c. Sound Classification

Conduct a study on how humans classify sounds and how humans react during different stages of classification. Example: A person who merely detects a sound (has heard it, but does not know what it is) may simply stop and orient his head in a manner to better hear the sound. A person who classifies the sound (has heard it and knows what produced the sound) may instantly react immediately.

d. Rule Library

Determine how Soldiers react to different sounds and create rules to add to the COMBAT^{XXI} "Rule Library." One example is improving the "communications" functionality by creating a rule that limits voice orders to the distance that a person can be heard shouting. The COMBAT^{XXI} developers already have a recording of shouting. The researcher would have to add the shouting sounds to the sound library, as described above, prior to creating the rule.

e. Spatial Orientation and Situational Awareness

Research how humans assimilate information gained from the senses of sight and hearing to build a mental map of objects around them in three-dimensional space (spatial orientation) and how they use this information to gain situational awareness. Possible questions to answer include: How does a human's mental map gained from vision and hearing differ from the actual situation? How do humans make decisions from their mental map? How do humans prioritize actions and reactions to the objects in the mental map? Applications include modeling entities that differentiate sounds emanating from known friendly entity locations (e.g. members of their own squad) from the same type of sounds emanating from an enemy location and Soldiers deciding to stop looking at a particular target in order to turn and locate the source of a noise that may be more threatening.

C. CONCLUSIONS

Although sound-detection algorithms for combat simulations have been in use since the early 1990s, these algorithms used only the distance between the sound source and the listener to calculate sound propagation, modeled a limited number of sounds, and provided perfect localization information. This thesis improves upon the previous work by giving the modeling and simulation community three tools: (1) A robust sound propagation and auditory-detection program, the ADM, which accounts for many environmental, listener, and sound factors. (2) A program, the AHAAH, that extracts the signature of any sound recording in a WAV file format. (3) Localization algorithms that are based on human experiments. These programs and algorithms were implemented in COMBAT^{XXI} and, through the face validation method, were found to improve the capabilities of the simulation by making it model human behavior more accurately. The combined capabilities of the ADM, AHAAH, localization algorithm and COMBAT^{XXI} provide modelers and analysts the tools to explore the concept of "every Soldier is a sensor" and the effects of hearing on a computer-generated combatant's ability to acquire targets, survive attacks, gather intelligence, and communicate with other entities, as well as to examine the use of acoustic sensors and non-lethal acoustic weapons on the battlefield.

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APPENDIX A. SOUND-LOCALIZATION EXPERIMENT DATA (SAMPLE)

Subject #	Trial #	Src #	Src Az	Src El	Rsp #	Rsp Az	Rsp El	RspTime	Hit	AngErr	dB Level
14	1	15	134.7	0	16	147	0	2.043	0	12.3	55
14	2	16	147	0	17	159.1	0	1.572	0	12.1	55
14	3	22	-134.7	0	22	-134.7	0	1.612	1	0	55
14	4	2	-58.3	0	1	-79.4	0	1.362	0	21.1	55
14	5	11	58.3	0	10	45.3	0	0.831	0	13	55
14	6	8	21.1	0	8	21.1	0	2.013	1	0	55
14	7	19	-172.4	0	19	-172.4	0	1.872	1	0	55
14	8	5	-20.9	0	5	-20.9	0	0.881	1	0	55
14	9	7	7.6	0	7	7.6	0	0.871	1	0	55
14	10	21	-147	0	21	-147	0	1.171	1	0	55
14	11	24	-100.6	0	24	-100.6	0	1.663	1	0	55
14	12	18	172.4	0	18	172.4	0	1.642	1	0	55
14	13	13	100.6	0	13	100.6	0	1.352	1	0	55
14	14	14	121.7	0	15	134.7	0	1.272	0	13	55
14	15	19	-172.4	0	19	-172.4	0	1.592	1	0	55
14	16	23	-121.7	0	22	-134.7	0	1.322	0	13	55
14	17	8	21.1	0	8	21.1	0	1.011	1	0	55
14	18	20	-159.1	0	20	-159.1	0	1.372	1	0	55
14	19	3	-45.3	0	3	-45.3	0	1.472	1	0	55
14	20	12	79.4	0	12	79.4	0	0.881	1	0	55
14	21	24	-100.6	0	24	-100.6	0	2.353	1	0	55
14	22	23	-121.7	0	22	-134.7	0	1.272	0	13	55
14	23	17	159.1	0	17	159.1	0	1.622	1	0	55
14	24	20	-159.1	0	20	-159.1	0	1.632	1	0	55
14	25	1	-79.4	0	1	-79.4	0	1.031	1	0	55
14	26	16	147	0	17	159.1	0	1.212	0	12.1	55
14	27	1	-79.4	0	1	-79.4	0	0.911	1	0	55
14	28	18	172.4	0	18	172.4	0	1.553	1	0	55
14	29	4	-32	0	4	-32	0	0.661	1	0	55
14	30	13	100.6	0	13	100.6	0	0.761	1	0	55
14	31	4	-32	0	4	-32	0	0.931	1	0	55

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APPENDIX B. COMBAT^{XXI} EXPERIMENT DATA

Scenario A					Scenario B				
NO Sound	SEM	SEP	SNM	SNP	NO Sound	SEM	SEP	SNM	SNP
9	1	1	0	1	10	1	1	1	1
10	2	2	1	2	12	1	2	3	4
8	0	0	0	2	13	1	1	1	1
10	1	1	2	2	12	0	0	0	0
11	2	1	1	1	12	0	0	0	0
10	2	0	0	0	8	0	4	1	1
11	1	2	0	2	12	1	3	0	0
10	1	2	0	0	12	0	0	0	0
10	2	4	1	0	12	2	1	1	0
10	0	0	0	1	12	3	3	4	5
10	1	1	2	3	12	3	3	1	1
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10	1	2	1	1	11	0	0	1	5
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11	0	3	0	2	12	2	0	2	0
10	0	2	1	0	4	1	1	1	0
10	5	1	3	1	12	1	1	0	1
11	5	0	1	0	5	3	2	2	2
10	0	1	1	1	8	0	0	0	0
10	1	1	0	3	12	3	0	0	0
11	1	2	2	2	12	0	4	2	3
10	2	1	2	0	12	1	1	0	0
10	3	2	2	2	12	1	1	1	1
7	4	2	1	0	12	4	4	3	3
4	1	3	3	1	12	0	0	0	1
9	0	3	3	1	12	2	1	0	0
10	0	1	0	1	10	1	3	1	1
11	2	3	0	1	12	0	2	1	1
9	1	4	0	0	12	1	1	1	1
10	1	1	0	0	12	4	4	1	1
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10	0	1	3	1	5	2	1	1	4
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11	0	6	4	0	12	0	0	0	0
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11	2	1	0	0	12	0	3	0	4
8	2	1	0	3	12	1	1	6	0
7	0	1	0	5	12	2	2	0	0

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9	0	2	0	0	8	0	0	1	1
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7	3	1	2	0	12	1	1	0	1
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8	1	1	1	1	12	0	0	0	0
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7	2	1	0	2	12	1	1	1	1
10	3	1	1	0	12	0	0	0	0

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